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the recording engineer

the broadcast engineer

2 to 8 the smaller recording studio

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About the Cover

Murray Allen of Chicago's Universal Studios looks over his new Synclavier. Read all about it on page 25.

About the 2-8trk Cover

Selcer Sound's control room features a Tascam M-512 mixing console, Otari 5050 MkIII 8-track recorder, and JBL and Auratone monitors. See page 16.

db, the Sound Engineering Magazine (ISSN 0011-7145) is published Bi-monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1986 by Sagamore Publishing Co., 1120 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions, $16.00 per year Canada) in U.S. funds. Single copies are $2.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office.

Letters

Dear Editor,

Today I received a copy of *dB* for the first time. I was delighted to read it. Although it is a techincal magazine, some articles are written in simple and easily understood language. I specifically liked the article by Larry Oppenheimer, "Making Sense Out of MIDI." It really helped me in putting things together.

By the time you read this letter, you should already have received my check and subscription form. If it's not too much of a bother, I would also like you to answer some questions for me if at all possible.

1. Where can I write to Larry Oppenheimer?

2. Can you please give the meanings of the following terms: SMPTE, THD, IM, EIA, CCIR, AES, and IEEE?

3. Where can I get Craig Anderton's "MIDI For Musicians" book?

Thank you very much in advance, and I am waiting for the answers in a future issue.

Sincerely yours,
Mike Shadowsky,
Tel Aviv, Israel

Of course it wouldn't be a bother, especially after you took the time to write such a nice letter. We'll answer your questions in the order you asked them.

1. You can write to Larry--or any other writer for that matter--in care of us. The address is:

dB, The Sound Engineering Magazine
1120 Old Country Road
Plainview, NY, 11803

2. SMPTE stands for the Society of Motion Picture and Television Engineers. The most common use of the letters is for SMPTE Time Code, a synchronizing system used to sync audio to film that was developed by the Society.

THD and IM are technical measurements used in product specifications. THD stands for Total Harmonic Distortion, and IM (sometimes written IMD) stands for Intermodulation Distortion.

EIA is the Electronic Industries Association.

CCIR is a European audio tape equalization standard.

The AES is the Audio Engineeing Society, and the IEEE is the Institute of Electrical and Electronics Engineers.

3. Craig's book is available from Mail Order Music, PO Box 572, Chester, NY, 10918. It retails for $14.95 (NY orders must add 8.25% sales tax). For domestic delivery, the company requests an additional $1.50 for shipping, so I would suggest you write to the company before sending them a check, since you would like shipment made to outside the US.

Editorial Calendar 1986-87

Synergetic Audio Concepts (SynAudCon) is sponsoring a series of two-day seminars dealing with solving audio and acoustic problems.

Demonstrations will include signal alignment, measurements of %ALcons and RASTI, the fundamental differences between impulse and energy time curve measurements, and how to design large loudspeaker arrays.

For more information , contact: Synergetic Audio Concepts, PO Box 669, San Juan Capistrano, CA 92693. Tel: 714-728-0245.

Anaheim,CA - February 3-4
Holiday Inn

Oakland, CA - February 11-12
Holiday Inn-Airport

Phoenix, AZ - February 18-19
Granada Camelhead Royale

Studio City, CA - March 3-4
Sportsmen's Lodge

The UCLA Extension Department of the Arts will offer two classes during the winter quarter dealing with sound for motion pictures, begining with a course taught by Brent Keast, manager, Cinesound Corp.,Inc. "Sound Recording for Motion Pictures," a 12-session class, will feature lectures, demonstrations and discussions. The topics will include basic physics of magnetic and optical recording, mixing and automated dialogue replacement.

Class meets at Cinesound,915 N. Highland, Los Angeles, on Wednesdays, 7-10 p.m., for a fee of $295.00.

Los Angeles,CA - January 7

The UCLA Extension will also have a class on "Sound Design of Special Effects for Motion Picture," which will be taught by Frank Serafine. This seminar will present an analysis and demonstration of effects and design. Class meets at Serafine FX, 438 Ashland Ave., Santa Monica, on Saturday, 10 am - 4 pm, for a fee of $140.00. For details, call the Department of the Arts, (213) 825-9064.

Santa Monica, CA - Febuary 14

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Circle 14 on Reader Service Card

Bruce Bartlett

Recording Techniques

SAMPLING, SEQUENCING, AND MIDI

Two new ways of recording music have been developed: sampling and sequencing. *Sampling* is recording a short segment of a sound and storing that sample in computer memory chips. *Sequencing* is storing a sequence of synthhesizer note parameters in memory chips. These kinds of recording are less common than tape recording, but they are important developments that recording engineers and musicians should be aware of.

First a definition. *Computer memory* is a group of integrated-circuit chips, each containing thousands of solid-state switches. Information is stored in binary format (1 = switch ON; 0 = switch OFF). These 1's and 0's are called *bits*, which stands for binary digits. Unlike a tape recorder, a computer memory has no mechanical moving parts.

Memory stores bits of information. Memory space is limited, and is measured in *bytes*, where 1 byte = 8 bits.

SAMPLING

Suppose you want to sample a live sound, such as a piano note, flute note, tom-tom hit, cymbal crash, or sound effect. You plug your recording microphone into a sampler or sampling keyboard, which records the sample as follows: The analog electrical waveform from the microphone is analyzed or sampled several thousand times a second and converted to digital data. This data is stored in random access memory (RAM), so that a digital recording is made (*Figure 1*).

Samples can be recorded from microphones, direct boxes, directly off synthesizers, off records, etc. The recording can be made first on an analog or digital recorder, then transferred to a sampling keyboard or external sampler.

Later, you play back the sampled sound by hitting keys on the

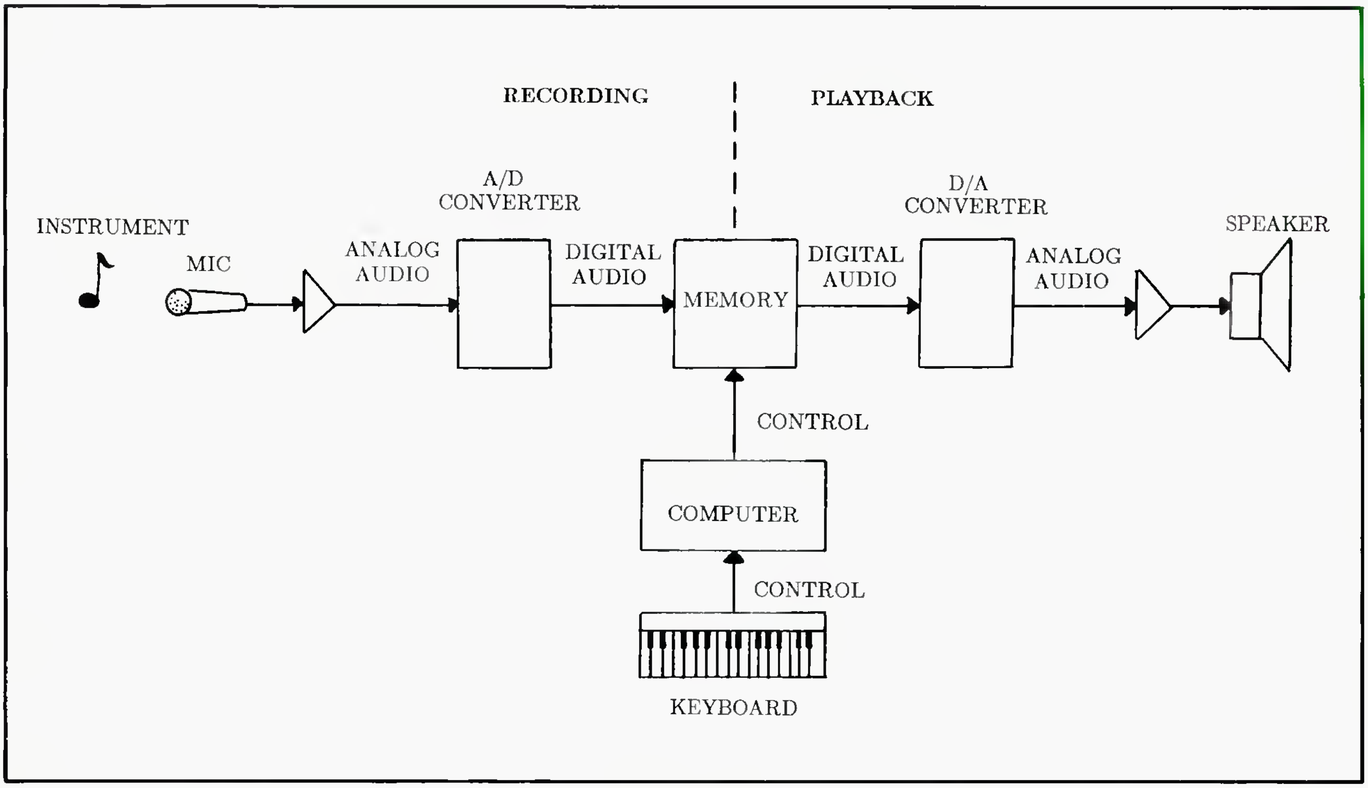

Figure 1. A simplified block diagram showing how a sampling device records and plays back sounds.

keyboard. The keypress triggers the sample. Which key you press determines the reproduced pitch of the sample. That is, different keys cause the digital information to play back at different rates, shifting the pitch of the sample.

Too much of this pitch shifting can cause an unnatural sound. Instead of having the entire keyboard control the pitch of one sample, it's best to record several

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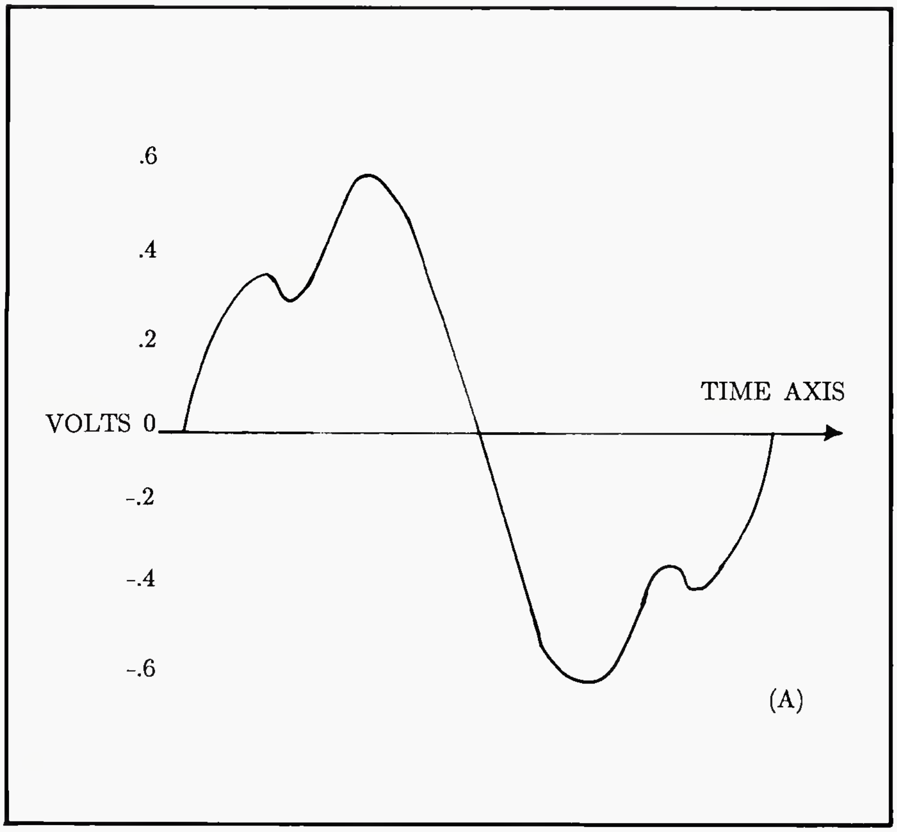
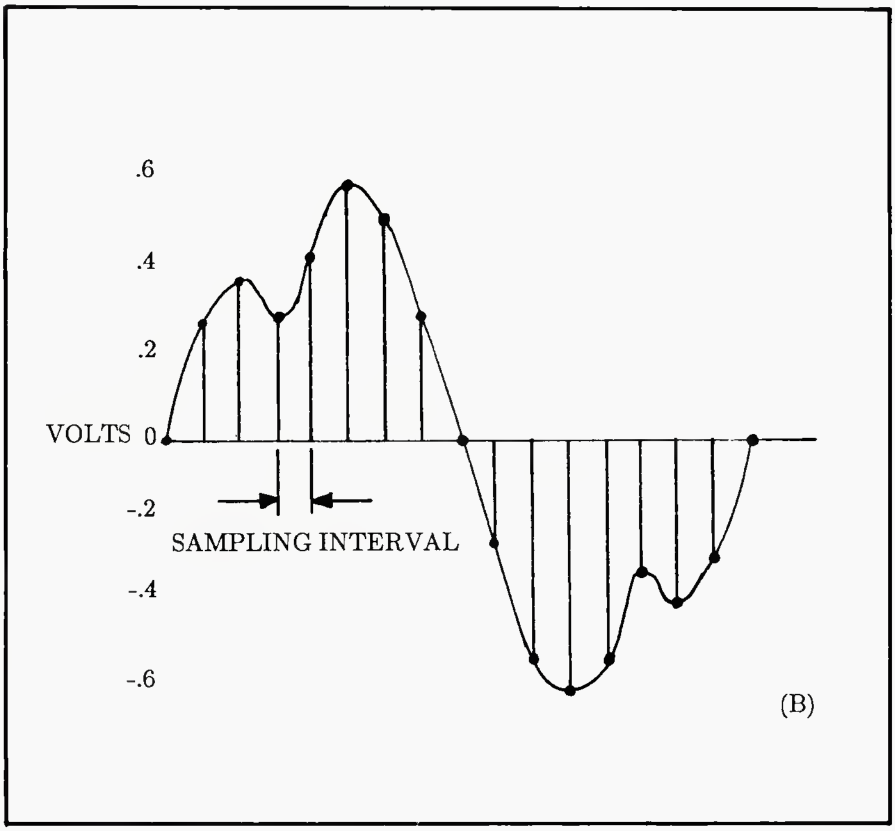


Figure 2.(A)The audio waveform enters the A/D converter.

samples at different frequencies–say one octave apart–and control the pitch of each of these samples within a smaller range.



(B)The voltage is measured or sampled at regular intervals.

Some keyboards and all drum machines have samples in permanent memory (ROM or Read Only Memory); these are digital recordings of real instruments stored in memory chips. The sample chips may be hard-wired or plugged in.

Many personal computers can be made to sample and store sounds. The software lets the user edit the sounds and control how they are played back. Sampled sounds can be saved on tape, floppy disk, hard disk, or–in the future –writable optical disks.

SAMPLING PARAMETERS

When an audio signal is sampled, it passes through an analog-to-digital (A/D) converter. This converter measures or samples the voltage of the audio waveform several thousand times a second. Each time the waveform is sampled, a binary number (made of 1's and 0's) is generated that represents the voltage of the waveform at the instant it is measured (*Figure 2*). These binary numbers are stored in memory.

The longer the binary number (the more bits), the greater the accuracy of the measurement. In other words, short binary numbers provide poor resolution of the waveform's amplitude; long binary numbers provide good resolution.

The *quantization rate* of a sampler is its amplitude resolution, measured in bits. The higher the quantization rate, the less the distortion and the greater the dynamic range. Commercial samplers range from 8 to 16 bits quantization. 8-bit is good, 12-bit is very good, and 16-bit is excellent.

The rate at which the waveform is sampled is called the *sampling rate*, measured in samples/sec. At a sampling rate of 40 kHz, 40,000 samples are generated for each second of sound.

The higher the sampling rate, the wider the frequency response of the recorded sound. The upper frequency limit is slightly less than half the sampling rate. If the sampling rate is, say, 20kHz; the sound you sampled will be reproduced up to about 9kHz. High-frequency sounds (cymbals) need a high sampling rate for fidelity; low-frequency sounds (bass, kick drum) can be recorded adequately with a low sampling rate.

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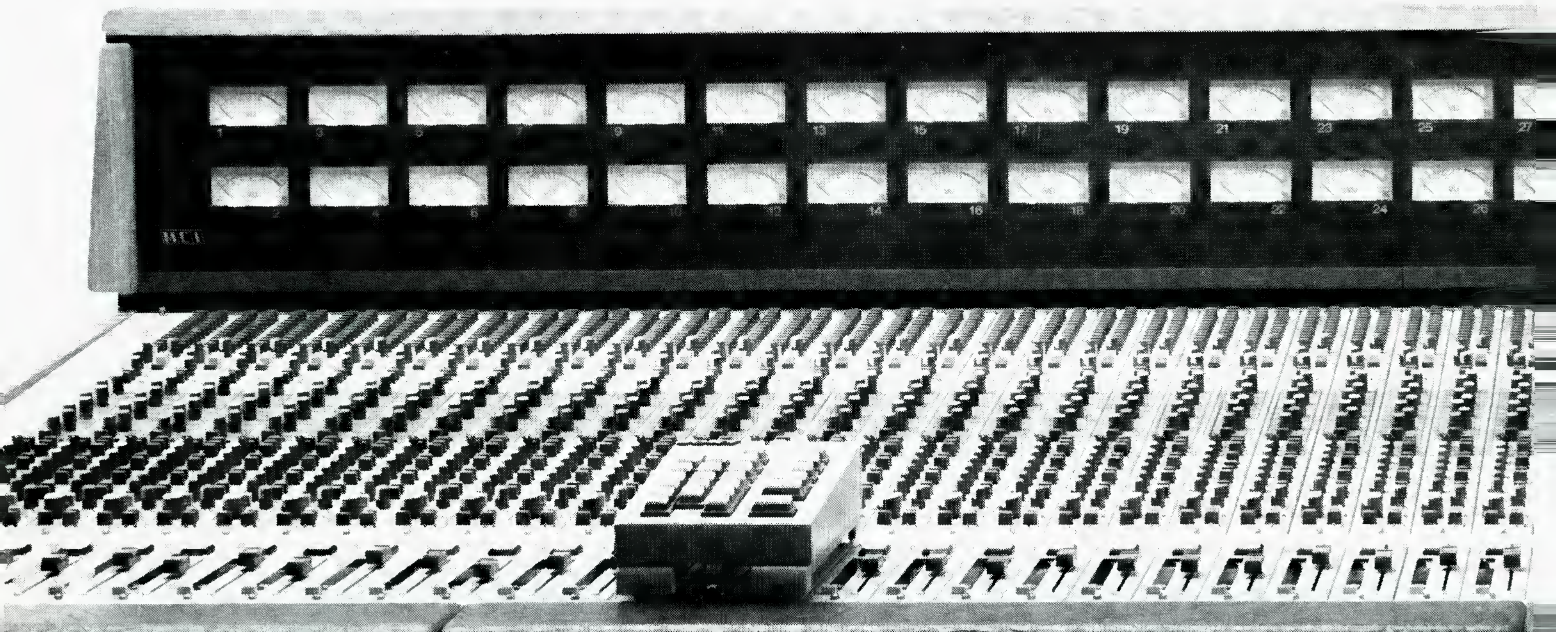

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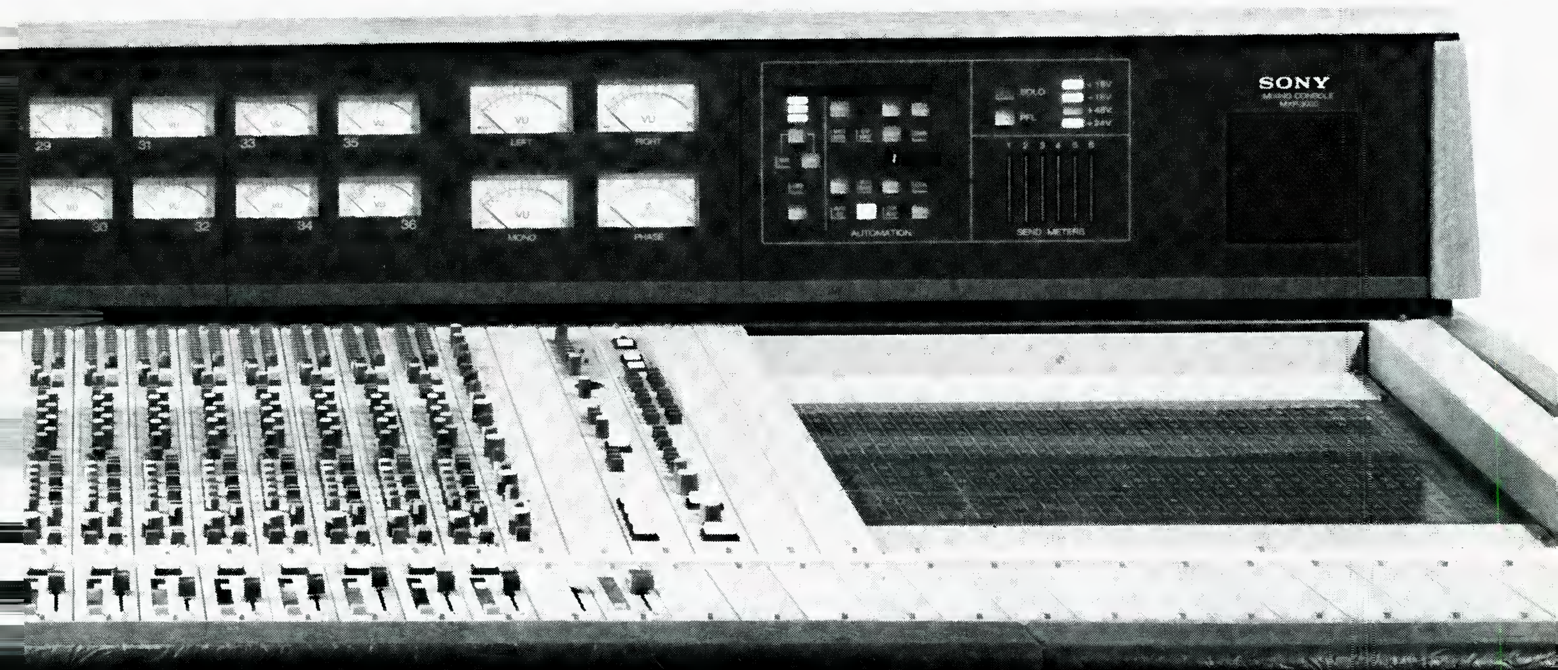

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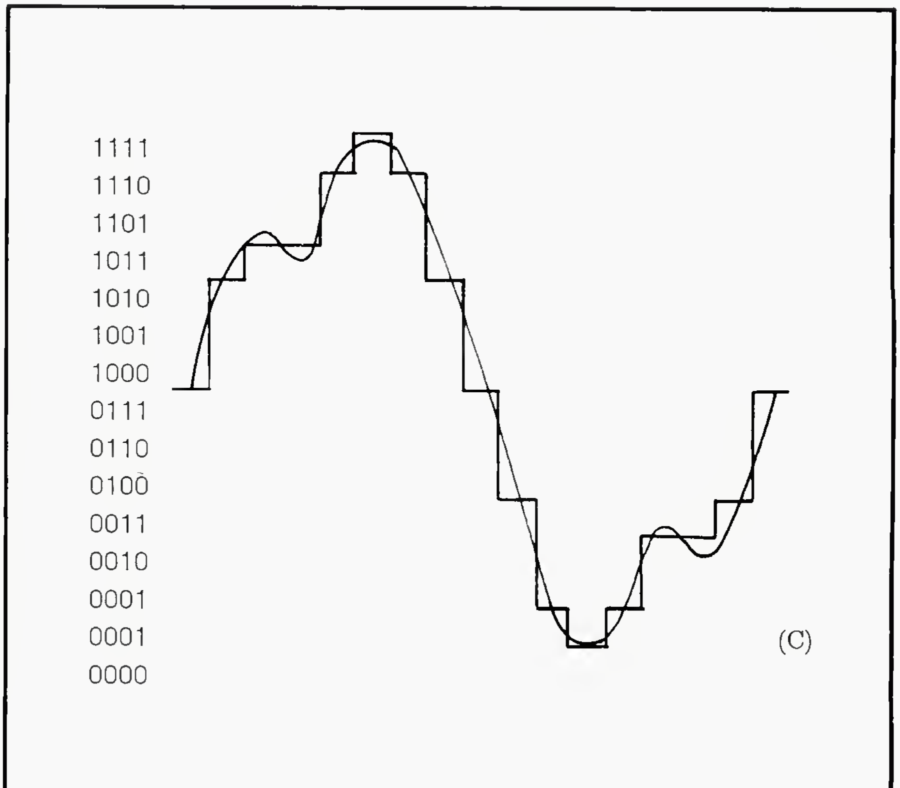
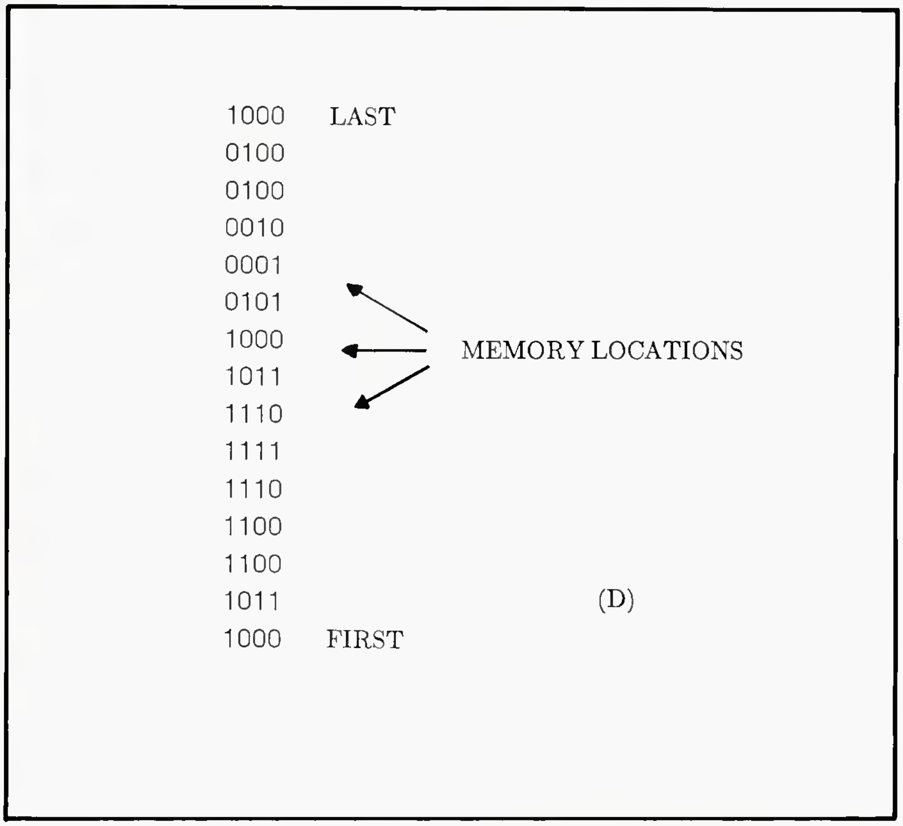


(C)The voltage measurements are quantized.

MEMORY CONSTRAINTS

As the A/D converter generates binary numbers, they are stored in memory. Each number goes to a separate memory location. Unfortunately, memory space is limited. Once it is filled, part of the recorded note is cut off. This puts constraints on the sample time, sampling rate, and quantization rate. The following equation shows how these four factors are related:



(D)The quantized values are stored in memory.

Bytes of memory filled by a sample =
Quantization rate x sampling rate x sample time

Stated another way,

Bytes = bytes/sample x samples/sec x sec

For example, if you have a sampler with 8-bit quantization (1 byte), and you set the sampling rate to 40 kHz, and record a 2-second sample, you use up 1 x 40,000 x 2 or 80 kilobytes of memory.

So, if you have a limited amount of memory, the sample has to be short, or the sampling rate has to be low, to avoid filling memory and cutting off the ends of notes.

Stated another way, the higher the sampling rate, the more memory is used up, because high sampling rates generate more binary numbers than low rates. To store the full duration of a note in a limited amount of memory, the sample must be relatively short. The higher the sampling rate, the shorter the sample must be. One-second samples are long enough

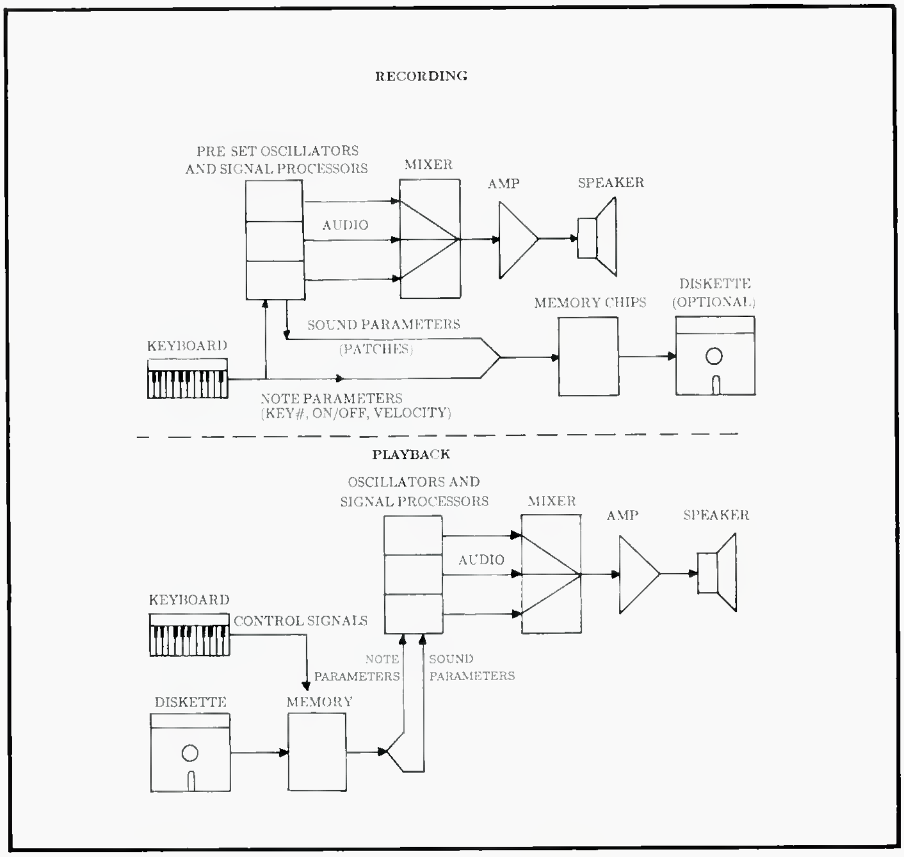


Figure 3. A simplified block diagram showing memory recording and playback of a sequence of synthesizer noted.

for the notes of many instruments, but cymbal crashes may require three seconds or more.

SAMPLING TECHNIQUES

When recording samples, first set the sampler to its highest sampling rate for maximum fidelity. If the end of the sound gets cut off (because memory is used up), try reducing the sampling rate.

If you fill part of memory with one sample, you can fill the rest of the memory with other samples, up to the memory limit. In other words, many sounds can be stored in different memory locations. This technique is called *multi-sampling.*

When you're multisampling, record high-frequency instruments or long-duration notes first, because they fill the most memory. If you record these sounds last, they may cut off if memory is used up.

Samples made for live performances should be already processed by effects boxes, but samples made for recording should be pure so that you can process them during mixdown. In either case, use a high-quality microphone, placed carefully, to get a clean and accurate sample.

You can edit the sampled waveform by using a computer connected to a port on the sampler. For example, you can remove or truncate silent portions of a sample to save memory. By trimming each sample as you make it, you'll get more samples into a given amount of memory.

SEQUENCING

With sequencing, you play notes on a synthesizer, and a memory chip stores which notes were played, their durations, and their sound settings. In other words, the memory records the note parameters, NOT the audio signal produced by the synthesizer. The sequence of notes you played, chords and melody, can be stored by a sequencer built into the synthesizer, or by an external computer running a sequencer program.

During playback, the sequencer activates the synthesizer. The parameters of each note are set and played according to what is stored in memory (see *Figure 3*). The synthesizer sounds are generated either from the synthesizer's oscillators, or from samples in memory.

In effect, it's a modern-day player piano. The instrument on which the original performance was played also reproduces the performance, with perfect fidelity.

For example, if you play middle-C on a synthesizer (261.63 Hz), the memory does not store a 261.63-Hz audio signal. Instead, it stores an indication that the middle-C key was pressed. Similarly, during reproduction, the memory does not play back a 256-Hz signal; rather, it triggers the oscillator that the middle-C key would play. That's how a sequencer records and reproduces music.

There are two ways of recording note parameters into memory: *real time* and *step time*. With real-time recording, you perform your music as you would play it on stage. The computer later plays back your music exactly as recorded. If desired, you can edit the piece. Step-time recording lets you enter notes one at a time at your own pace. The music plays back at a normal tempo.

Many synthesizers can store sound parameters (patches) or rhythm data on cassette tape or magnetic disk. The data can be loaded back into the instrument to recall rhythms or patches. You can even buy pre-recorded cassettes that contain sounds programmed by professional musicians.

SUMMARY

In review, you create sounds by recording them into memory (sampling), by using pre-recorded samples, or by generating sounds with a synthesizer. Once you have these sounds, you play them with the keyboard.

A sequencer or external computer can record what you're playing. The memory stores the note parameters (key number, duration, etc.), not the audio signals. At the touch of a button, the sequencer will automatically reproduce the notes you played by telling the synthesizer what notes to play. The sounds of these notes are generated either from stored samples or from the synthesizer oscillators.

MEMORY MULTI-TRACKING

Now let's say you've sampled many sounds, and have built up a *library* of these sounds or voices. All these different voices can be

stored and called up at will.

Suppose you play a sequence of notes and record the sequence into computer memory. This sequence might be drum beats, a bass line, chords, or melody. A sequence of notes of one voice is called a *track.*

After recording a bass line, you can go back to the start of the sequence, play the bass track, and add a flute part in sync with the bass line. The flute melody or sequence is stored separately in memory. Then you can go back to the top and add drums. It's just like overdubbing with a multi-track tape machine, except there's no tape to rewind and no generation loss. You can punch in/out and mix down these tracks just as in multitrack tape recording.

If you have ever played with a drum machine, you know how synthesizer overdubbing works. You play, for example, a four-bar riff on the hi-hat and kick drum keys. This riff is stored in memory. Then you can play it back while adding a tom-tom fill. That combination is stored. Then you can add a cowbell, and so on. The recording can be mixed by adjusting the faders on the drum machine for each instrument.

SYNCHRONIZING SYNTHESIZERS WITH MIDI

So far we've discussed recording with a single synthesizer, but there are more possibilities. Several synthesizers can be synched together to produce the effect of a band playing. You might have two or three synths and a drum machine synchronized and playing all at once.

Several synths can be linked to a drum machine, which provides the basic pulse that sets the tempo. Or they can be connected to a click-track machine which generates a timing pulse.

These memory recordings can be played back during live concerts. In this way, synthesizer musicians can play note-perfect performances every time. Or they can override the sequence and play manually to react to the other musicians' playing.

The system used for interconnecting synthesizers is a MIDI interface. Let's explain what MIDI is.

MIDI stands for Musical Instrument Digital Interface. It is a specification for a computer interface that enables several different brands of instruments to be connected together. That is, it permits electronic musical instruments and computers to communicate with each other through a standard cable.

With the MIDI connection, several MIDI-equipped synthesizers or drum machines can be controlled from a single keyboard, guitar, microphone, or drum machine. Similarly, a computer program can control several keyboards, drum machines, and effects devices. A program can perform an automated mixdown on certain consoles by remembering and re-setting console control settings.

With MIDI, you can make an electric guitar sound like a flute, or any other instrument. Here's how: You play the guitar, which is patched to an audio-to-MIDI converter. The MIDI signal controls a synthesizer. The synthesizer follows the note envelopes and fundamental frequencies you're playing on the guitar. If the synthesizer is set to play a flute sound, and you pluck a guitar string, you hear a flute!

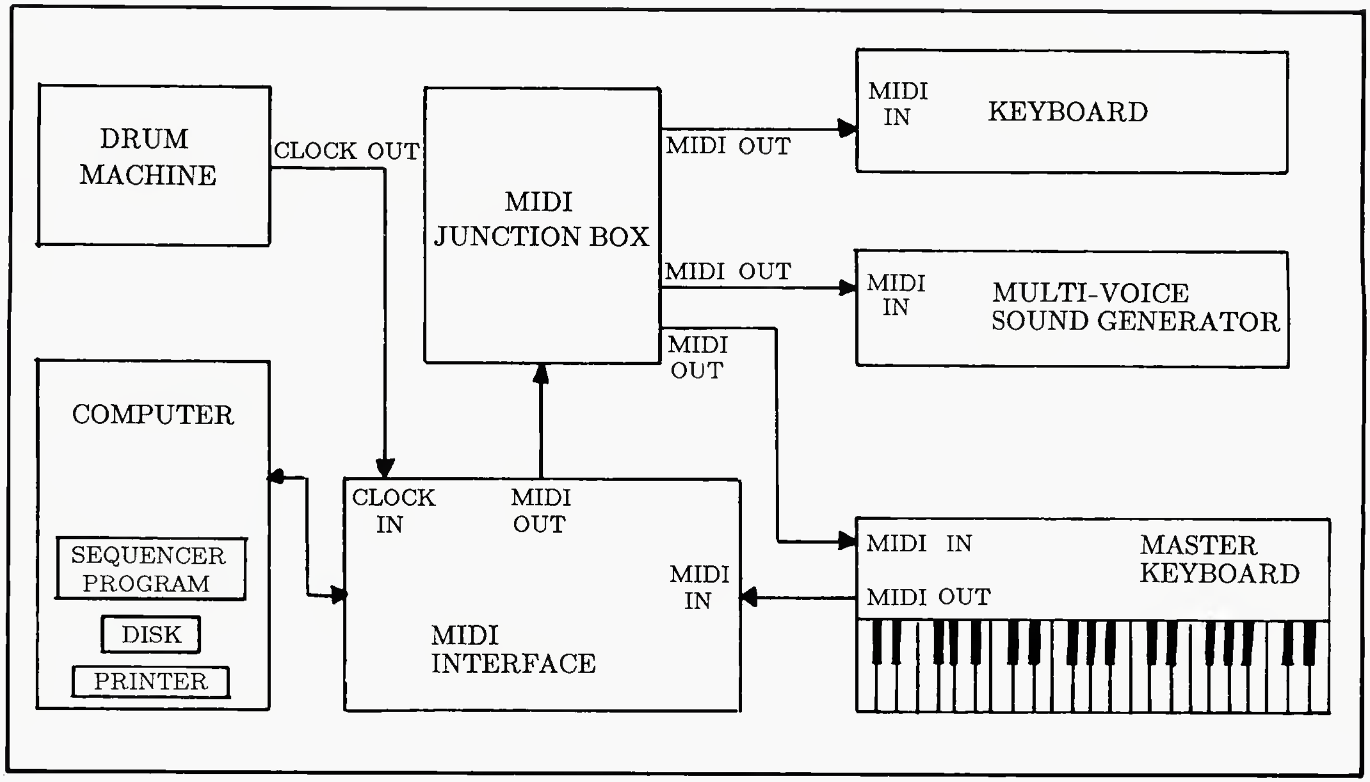


Figure 4. A typical MIDI studio.

Up to 16 channels of digital data (31.25 kbaud) are sent serially through a single MIDI patch cord. Each channel transmits information that controls a MIDI instrument. Several instruments can be connected so that one MIDI channel controls all those instruments simultaneously. Musicians can layer several sounds in one pass, as well as recording each instrument separately.

The MIDI cable is standardized with a 5-pin DIN connector, but each manufacturer applies the MIDI spec in a different way. Not all MIDI instruments are compatible; for example, they may have different functions controlled by the same channel. Fortunately, many instruments can be reliably interfaced with MIDI accessories.

A TAPELESS STUDIO

With sampling keyboards, synthesizers, MIDI, and memory multitracking, you can set up a complete recording studio that operates without tape, and without live musical instruments!

Conventional recording studios have several microphones for picking up instruments and vocals. A tapeless studio might have only one high-quality mic that is used to sample various instruments one-at-a-time into the sampling keyboard.

Conventional studios also have multitrack tape machines to record several instruments, each on its own track. Tapeless studios can do without the multitrack tape machine because the keyboard contains a multitrack memory recorder (sequencer). Or you can use software that converts a personal computer to a multitrack recorder.

Figure 4 shows a typical MIDI studio. It includes a personal computer, drum machine, keyboards, multi-voice sound generator, and interfaces. Here's how it works:

Using sequencing sofware, the computer acts as a multitrack recorder or sequencer. A MIDI interface lets the computer talk MIDI language to the rest of the system.

The clock output of the drum machine drives the computer-sequencer, which in turn controls the keyboards and sound generator through a MIDI junction box. This box feeds MIDI signals to all the keyboards simultaneously. This arrangement is better than a daisy-chain connection, which slows the transfer of MIDI data.

The master keyboard controls all the others. It is plugged into the MIDI interface and the MIDI junction box.

After all the tracks are recorded, the computer-sequencer plays the multitrack recording, which activates all the keyboards and the sound generator. The outputs of the keyboards, sound generator, and drum machine are mixed through a console (not shown) onto a 2-track tape deck.

What if you want to add vocals to the mix, but the vocal track is too long to sample? You could record the instrumental mix on a multitrack tape recorder, then overdub the vocal on another track. Alternatively, some expensive keyboards let you digitally record the vocal track onto a floppy-disk recorder.

CONCLUSION

We've described two ways of recording music into computer memory: sampling (digitally recording short sounds) and sequencing (storing sequences of note parameters). With memory multitracking, several different instruments can be played, recorded, overdubbed, and mixed with a single keyboard. MIDI connections enable a computer program to play several keyboards, and permit musical instruments to sound like other instruments.

Clearly, the marriage of computers and music is bearing amazing offspring. This is the future of recording. ■

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2 to 8 trk

Personal Sound

Ken Selcer and Craig Lambert

Here's a fine small studio located in downtown Boston. We present this as typical of what is happening around the country.

When I started Selcer Sound two years ago, I was out of a job, had little money and was a frustrated guitarist with slim commercial prospects. The only assets I brought to the project were five weeks of training in audio recording, twenty years of playing music, and a degree in music from Berklee College of Music in Boston. Perhaps my greatest asset was a strong intention to make my contribution to the music industry.

Today, I own and operate an 8-track recording and mixing facility in Boston. I opened my doors fifteen months ago after travelling a long, hard road in setting up the studio.

At the beginning, I knew very little about setting up either a recording studio or a business of any type. Therefore, I did an enormous amount of research. I talked to everyone I knew in the music and audio fields–equipment dealers, engineers and musicians. I combed magazine racks and libraries for information. I wrote a business plan and got the necessary contracts drafted.

The fact that I had done my homework was apparent to friends and family members, who lent me the money to get started. My fund raising efforts took only one week! I was on my way!

THE EQUIPMENT AND THE BUDGET

Since I wanted the best equipment, but was confined to a $15,000 budget, I faced a dilemma. My solution was to buy the equipment via mail order, rather from local dealers. This enabled me to acquire the most versatile components without compromising audio quality. For example, I chose an Otari 1/2-inch MK 5050 8-track recorder. The extra outlay for that machine proved to be a smart investment; knowledgeable musicians are always impressed when they learn that I have the Otari, and its well-earned reputation helps to draw business into my studio.

I settled on a Tascam M-512 12x8x2 mixing console, equipped with eight buses, which eliminates the need for special patching. The time savings outweigh the additional cost. For this reason, I chose a 12x8x2 console instead of a 16x4x2; I have found that 12 inputs can handle virtually any session I book. I use five or six channels for drums, one each for bass and guitar, perhaps another input for keyboard and one for scratch vocals. This still leaves me with a few inputs free for effects. The Tascam has proven itself to be extremely clean sounding, versatile and easy to use.

Another piece of quality equipment that draws musicians into my studio is the Lexicon PCM-60 digital reverb. I chose this reverb for its client-drawing power as well as its beautiful sound. People love it when I mention the Lexicon. "I've looked all

Ken Selcer owns and operates Selcer Sound in Boston, MA. Craig Lambert is a Boston-based writer.

Figure 1. Ken Selcer working at his Tascam console at Selcer Sound.

over town for a studio that I could afford to book which had that PCM-60," said one guitarist.

I also have a Symetrix 522 compressor/limiter/noise gate/expander. It has two channels, and its operation is straightforward. If a singer wants to sing on the basic tracks, the expander feature allows me to isolate vocals from the rest of the room sound,

Other equipment in my studio includes a Delta-Lab CompuEffectron CU-1700 (digital delay, chorus, flange, etc.), JBL 4312 and Aurotone T-5 monitors, QSC 1400 power amplifier, and Tascam 2- and 4-track recorders. In general, I selected my equipment for its reputation, ease of use and versatility. I'm pleased to report that, after over a year in business, this array of hardware has met both my budgetary and audio-engineering needs.

A PLACE FOR THE STUDIO

I negotiated an agreement with Home, Inc., a non-profit arts organization in Boston, for the use of rooms in their building at 731 Harrison Avenue, near Boston City Hospital (the facade of which appears on the St. Elsewhere television series). Friends helped me build and set up my control room, either donating their time or bartering it in exchange for my services--future studio hours, for example.

The dimensions of the recording room, which I share with Home, Inc., are 25x24x12 feet. Since it doubles as a video production studio and has a good, live sound without major acoustic problems, I needed to make only minor modifications to the room. I built baffles for isolation of instruments, applied some tectum to the ceiling and walls, and distributed other baffles around the room. I was ready to book my first session.

STUDIO PROMOTION

Selcer Sound's advertising campaign was already underway by the time I opened my doors in July,

1985. My market, at that time, was rock musicians who were "putting the pieces together"–i.e., getting their acts ready for the marketplace. I mailed out lots of flyers, placed a few ads in local music papers, and spread the word to everyone I knew personally. Slowly, business began to come in.

Most of the clients I've served over the past fifteen months have been people on the verge of breaking into the Boston music scene, or having recently broken into it. They want to record demos for getting club dates, for courting management, for their friends and themselves, and for radio airplay. Large studios are too costly for their needs, and they feel more comfortable in a 8-track room anyway. They need audio professionals who have the time and energy to work with them on their projects, and to help them develop in their careers. The city brims with students and young musicians of this type who are continually forming new bands and who want high-quality, professional recordings made at a reasonable price.

I designed my studio to meet the needs of such up-and-coming local musicians. My clients not only shop for the best sound available, but also seek to learn about recording techniques and have some fun in the process. Selcer Sound maintains a relaxed studio atmosphere which facilitates such personal and artisitic goals. The high-tech equipment and steep hourly rates of the larger studios can be an intimidating presence.

Being a musician myself helps me to relate well to clients in creative areas. I know what they are doing and what effects they are striving for, and can discuss options from the artist's standpoint as well as the engineer's. Clients have often remarked that my studio's receptive attitude helps to unlock their creativity.

A LOOK TOWARD THE FUTURE

As my business matures, I acquire new pieces of hardware to handle the situations that arise, rather than buying every new gadget that appears in the marketplace. For example, experience has shown me that two channels of compression are not enough. I use compression on the kick drum or snare, and I find that I also need it for the bass guitar on most sessions. Therefore, my next equipment purchase will be a Symetrix 522, giving me two additional channels.

Since I record at extremely *hot* levels, I haven't yet had a real need for noise reduction, and thus far, have not acquired such equipment. I do plan to upgrade to better microphone stands with booms, since there are many spatial configurations that I have trouble mic'ing. I also plan to add a keyboard, an acoustic piano, a good drum machine, and a bass guitar. Clients have requested all of these instruments at various times. I intend to replace my open-air headphones, which allow too much leakage onto the tracks when singing overdubs. My microphones include Sennheiser 421's, Shure SM-57's, AKG D-1000's, and Electro-Voice RE-20's and PL-95's.

TECHNIQUES

For some time, I have been grappling with the problem of isolating instruments from each other. With no islolation booths, close-mic'ing techniques are imperative. During overdubs, I can add ambient sound with reverb and/or delay. I baffle the drum kit, close-mic the bass (or go direct), and position the bass and guitar on opposite sides of the room. Generally, this solves the leakage problem.

Having no window between the control room and the recording studio, I also had to find a way to communicate with the musicians. The answer was a video camera fitted with a wide-angle lens in the recording room, connected to a monitor in the control room. In combination with my headphone system, this arrangement solved my communication needs.

I have also done voice-over recording for lecture tapes, training films, demos for commercials and commercials themselves. Boston has a lively market for voice-over work, and I plan to expand my activity in this area, which allows billing at corporate rates which are higher than those I charge individual clients. Voice-overs are relatively easy recordings to make, and can become a lucrative line of business for any studio.

Figure 2. Tascam 22-4 4-track and 22-2 2-track tape decks are side-by-side on a small rack which is housing a patch bay, Lexicon PCM-60, Symetrix compressor/limiter/noise gate/expander, and DeltaLab CompuEffectron.

My goal is to create the capabilities for a wide variety of recording projects, from folk to rock to jazz to commercials and training films. Boston's audio recording scene is highly diversified, and as I become better-known in different areas, more business comes my way. I mail out flyers to musicians and to faculty at music schools and advertise in local newspapers. I give out business cards and speak to everyone about my studio. Word of mouth is still the best advertising; a satisfied client not only returns, but brings in new business.

Selcer Sound also engages in other recording-related activities. I do live sound for bands and single artists, produce and promote small local concerts and benifits, and work on live video shots. My Tascam M-30 and other portable equipment from my studio handles live recording and small live sound shows.

Selcer Sound is evolving into a service/resourse center, a multi-purpose business that offers an alternative to high-cost, intimidating, large-studio operations. Mine is a personal but professional studio, one which brings a low-key, creative approach to a comprehensive array of music, recording and production services. ■

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On Taxes

Mark E. Battersby

Asset Dispositions

Investors in stocks, bonds or real estate enjoy a tax advantage that is difficult for home studio owners or sound engineers to duplicate. Upon the sale of investment property the resulting gain may qualify for treatment as a capital gain and be taxed at a special low tax rate. Fortunately, properly handled, the sale of the assets used in a recording activity may also qualify for the beneficial capital gains tax rates–including a home or professional recording studio.

Profiting from those low capital gain rates is, unfortunately, not always an easy matter, particularly under our complex tax laws. Take for example, the ordinary sound equipment that forms such an integral part of the activity.

Under prescribed conditions, the benefit of the long-term capital gain provisions is applied to the disposition of certain non-capital assets held for longer than six months (one year for property acquired after 1987). This special treatment also applies to the sale or exchange of some property used in a trade or business or in a so called *for profit* transaction. The IRS' special rules also encompass the destruction of property, theft, seizure, requisition or condemnation.

The phrase *property used in the trade or business* means only what the International Revenue Service refers to as personal property used in a trade or business and subject to depreciation or cost recovery allowances. Real property (such as the lease-hold of land and improvements) used in a trade or business activity also qualifies.

If the property is not actually used in the trade or business of the home studio, but is held for sale by a dealer in such property, it cannot qualify for the preferential tax treatment. Examples would be real estate held for sale by a real estate dealer or sound equipment held for sale by a musical instrument dealer.

When property is partly business and partly non-business, the business portion is subject to this unique Code Section 1231 treatment; the non-business portion is usually already a capital asset, in any case.

The Code Section 1231 rules apply to fully depreciated assets because these retain the character of assets subject to depreciation even though no further deduction is allowable. Although they are still carried on the books at the scrap or salvage value used when originally computing depreciation and even though they are no longer actually being used in the sound or recording business, they are not converted into scrap.

If the gains from these transactions exceed the losses from such transactions, that is, if there is a net gain, then each gain is a long term capital gain and each loss is a long term capital loss. If on the other hand, there is a net loss, then each loss is an ordinary loss and each gain is ordinary, fully-taxable income.

Quite simply, long term capital gains are offset against long-term capital losses. If there are excess long term capital losses, only a relatively small amount of those losses may be used to offset income from other sources. Ordinary losses, on the other hand, can be used in their entirety to offset other income. The desirability of the low capital gain tax rates and the limit on long term losses makes long term gain and short term losses the goal to aim for.

Beginning in the 1985 tax year, if the netting of Code Section 1231 gains and losses from the sales of business assets produces a net Code Section 1231 gain for the year, a five year look back rule applies to re-characterize the current year's net Code Section 1231 gain as ordinary income to the extent of the net Code Section 1231 losses in the look back period.

Remember, however, that this *look back* provision does not affect the losses incurred by your home recording or sound activity during the previous five years. Rather, it converts the Code Section 1231 gains for the present tax year into ordinary, fully-taxable income to the extent of Code Section 1231 losses reported on earlier income tax returns.

The complexity of this particular section of our tax law, the Internal Revenue Code, extends to gains and losses from involuntary conversions (condemnation, destruction or loss by theft or casualty). Gains or losses from these involuntary conversions generally qualify as Code Section 1231 assets except when the owner is required or chooses not to have the gain recognized. Conversion of both property used in a trade or business and of cap-

ital assets held in connection with a trade or business and of capital assets held in connection with a trade or business or in a for-profit transaction are included--but only if the property has been held for the required time period (six or twelve months).

Gain or loss from the sale or exchange of land used in a trade or business and held for more than six months generally qualifies for this unique Code Section 1231 treatment. However, gain or loss from the sale or exchange of land held by a studio owner primarily for sale to customers in the ordinary course of his trade or business is not subject to Code Section 1231 treatment.

An individual need not be a professional salesperson or developer in order to be treated as a *dealer* for tax purposes. In fact, according to the US Tax Court, gains from the sale of land that had been a part of one taxpayer's family farm before it was subdivided into housing lots were taxable as ordinary income rather than as capital gain.

The taxpayer in this situation had argued that he was forced to sell the land in order to support his elderly mother and that subdividing the land into lots made it easier to sell. Unfortunately, however, the Tax Court found that he had purchased the land from his mother to sell at a profit, had never intended to hold it as a long-term investment and had devoted a substantial amount of time, skill, and financial resources to improving, developing and selling property. Consequently, the court ruled that the taxpayer was actually engaged in the business of sub-dividing real estate for sale and the land was sold in the ordinary course of his *business.*

For the average home studio owner, the question arises whether land or improved realty is held primarily for sale to customers in the ordinary course of a business is one of fact. As an aid to deciding this question, the courts tend to look at a number of factors such as:

The purpose or reason for the taxpayer's acquisition and disposal of the property;

The continuity of sales or seller-related activity over a period of time;

The number and frequency of sales;

The extent to which the taxpayer or his agents engaged in sales activities by developing or improving the property, soliciting customers or advertising;

The substantiality of sales when compared to other sources of the taxpayer's income;

The desire to liquidate unexpectedly obtained land holdings (such as by inheritance); and overall reluctance to sell the property and the length of time the property was held.

Utilizing Code Section 1231 to minimize the tax bite which usually results when business assets are sold or exchanged makes a great deal of sense--despite the complexity of the rules. Plus, with Code Section 1231, it is not necessary that there be both gains and losses. If there are only gains or if there is only one Code Section 1231 transaction (resulting in a gain), the losses are treated as zero, and, in such cases, there is an excess of Code Section 1231 gain over Code Section 1231 losses.

The bottom line result is a capital gain, the same capital gain used by investors to shelter a portion of their profits from the bite of the tax collector. Those same capital gains can be achieved if the hobbyist sells a keyboard or any other capital asset which meets the holding period and other requirements for capital gains treatment.

At the present time, the capital gain tax rate is a maximum of twenty percent. On the annual income tax return only forty percent of net capital gains are considered as taxable income. Thus, with a maximum tax rate of fifty percent the effective real tax rate on capital gains can never be more than twenty percent...little wonder why so many astute investors maneuver to qualify income as *capital gain.*

Will your next sale qualify for the preferential capital gains treatment? If you're worried about losing that tax benefit because frequent sales may label you a dealer, stay tuned, we'll delve into that sweep in the near future. ■

Brian Battles

Ad Ventures

Since you are probably a recording expert, or at least an ardent hobbyist, I imagine you have no trouble gaining access to technical improvement literature. For instance, this very magazine is chock full of articles describing techniques, construction projects, solutions to problems, and so on. Yet, how often do you look to a recording industry publication for sheer inspiration? Most successful members of this industry consider sound engineering to be an art as well as a science, and since I prefer to view myself more as an artist than a scientist, this month I will address some of the more abstract concepts involved in your commercial endeavors. Sometimes it's preferable to have a *psych up* rather than a *how to* session.

In most cases it is more important to *want to* succeed than just to know how to do things. For example, everyone seems to know someone who appears extraordinarily gifted, creative, perhaps ingenious, yet cannot hold a job, finish a project, or even go one full day having accomplished something of real value. But that's okay; it takes all kinds of people to make up the world, and that makes your job easier because most other folks won't be competing against you. Of all the people you know who operate recording businesses, how many are goal oriented and driven to success? If you think about it, the odds are on your side.

To paraphrase the last words of the late Jimmy Hoffa, I would like to jump right into the concrete aspects. These are ideas that you can put to work right now. Before you plan to use your recording studio to produce radio and television commercials, you must consider yourself a professional. I know that the term is overused, since technically as long as you get paid for doing something you are a *professional*, but let's use the word in its more common connotation; a professional is an individual who is skillful, diplomatic, classy, and takes enormous pride in the excellence of his work. In order to place yourself in the position to properly apply those adjectives to yourself, you must possess one vital trait: self discipline. Be organized! I think you will agree that most people who fail tend to be rather disorganized, confused, sloppy, unreliable, and often miss out on promising opportunities.

Just about everybody can learn discipline, but it is hard. I won't try to fool you, there are no real shortcuts to take in the application of discipline. That is why most people are not as successful as they could be. In the recording industry there are a lot of misconceptions, particularly when it comes to business. The vast majority of us became involved in this field because we either love music or love gadgets. Face it, you are included, aren't you? It's nothing to be ashamed of, but you truly must understand that love of your profession, healthy and desirable as that may be, is not the only prerequisite. Just as a rowboat needs two oars to move rapidly forward, you need a well-balanced supply of strengths: ability (combining talent and passion) and discipline (organization and commitment).

Since you are reading this, I assume you already possess the ability part. But if you are like almost everybody you meet, you lack some discipline. Let's examine this more closely: are you compulsive about things like equipment maintenance? Do you actually clean and demagnetize tape heads and guides, calibrate all controls, lubricate parts, replace worn belts and pinch rollers, scrape off excess editing pencil wax, constantly use fresh razor blades, and vacuum your studio and control room all the time? Every single one of those chores has a definite effect on your product's quality.

All right, let's say you are a Felix Unger in the studio. How about these: Do you write down all appointments, plans, schedules, events, and important notes in a single book that you carry wherever you go? Do you send out thank you notes to clients, prospects, suppliers who give you discounts or good service, and others who make your life a bit easier? (I see less hands going up now.) Okay, let's look at really boring stuff, such as finances and accounting: Are this year's business related receipts all safely filed in a convenient location? Are they all dated and marked with brief explanatory notes so you will know what you spent and where? This is not just for tax purposes. Many smart businessmen like to look at records to help decide on favored sources of equipment and services, and it helps to know how much you spend with someone when you

want to negotiate for better treatment.

I can tell that you are starting to feel doubtful of your level of discipline, but I want to be ruthless; let's blow off the remaining sinners. Are you always prompt in paying bills? A good credit rating is critical in business, especially since recording equipment can cost so much. Someday a Master Card could come in quite handy when your local audio shop puts your dream mixer on sale and your bank account is as empty as Pee Wee Herman's head.

How about meeting clients: Are you always well-groomed, neatly dressed, sufficiently prepared? I never go to a business meeting, no matter how informal, unless I am carrying a few essentials such as my appointment calendar, a note pad, pens, business cards, and so forth. I prefer a briefcase, but even a nice little portfolio folder will do for some situations. Believe it or not, a container of breath mints or Binaca in your pocket can do wonders (at least for your confidence). I can never pay close attention to someone with bad breath, or too much cologne or perfume. It also helps to get used to wearing nice slacks instead of jeans, dress shirts instead of T-shirts, loafers instead of jogging shoes. Certainly you must establish your own dress code for conducting business activities, but you will find that when in doubt, you should slightly overdress. For all this does is make you look more serious and attractive. It really is not a hard habit to learn.

To continue with discipline, let's look at your work environment. Each and every one of your tapes should be impeccably marked, filed and catalogued. Your customer files must be neat and up to date. Splicing supplies, tools, accessories, cables, and other odds and ends have to be judiciously returned to their storage spaces after each use. Machinery is easy to keep in good shape if you set up a simple maintenance log, and all equipment documentation, manuals, schematics, brochures, warranties, receipts, and reference materials need a permanent home.

As your business starts to grow, the need to manage your time as effectively as possible starts to increase proportionately. List tasks and appointments each day or week in order of priority, and you stick to your list diligently. Nobody likes to do unpleasant jobs, but they are the ones that give you the most satisfaction once hey are completed. Start with the tasks you dislike most. That way, as the day progresses, your work will seem to get easier and more enjoyable.

My memory is horrible so rather than bother trying (and failing) to remember the stuff I must get done, I use a system which I consider invaluable. It's my "daily recall" file. I have a desk drawer full of hanging legal file folders, each marked with a number from one to thirty-one. The first thing I do in the morning is reach into the front folder (today's) and pull out the contents. Then I place today's folder in the back of the drawer so that tomorrow's is now in front. Now I can go through the notes, papers, and duties to be performed today, plan my schedule to mesh with the entries in my appointment book, and know I'm not missing anything. As my day continues I write down things to be acted upon at a future date, mark them to go in the appropriate folder, and file them away. At one quick glance I can look into my recall drawer and get a fairly accurate estimate of how many things I have to do on a given day. There are, of course, many variations on this system, and I certainly don't claim to have invented it, but it is a method that works very well, and it has made a huge change in my life ever since the day I initiated it.

Punctuality is a true mark of a professional. Show up on time for appointments or even early. Finish editing a tape a few days before you promised it. Confirm meetings in advance by phone, and then reconfirm them. Send out the check you promised. Get to the post office before it closes today. Call that guy back at 8:55 AM if you said you would phone at nine. Think of all the chores and activities you must accomplish and set definite deadlines. Make them realistic so you can fulfill them. Give people more than they hoped for, sooner than they expect, and be consistent and pleasant, no matter how unimportant it might seem at the moment. Remember that you have no real perspective on today. Nothing will be fully clear until it is in the past, and by then it is too late to change. Life doesn't allow you to go back and *fix it in the mix.*

I can give you a true life example of how gritting one's teeth and providing superb service can reward you with extraordinary dividends. One of my clients who owned a small discount stereo shop was a particular pain in the nether cheeks. He always needed to revise a finished commercial three or four times before he would give it his grudging approval. I had eagerly signed up to work for him back when I was first getting into freelance commercial production, so I did not charge him a very high rate. He only bought a few spots from me each year, and he often paid me late or made it difficult to contact him for meetings. I could have dumped him. Had I performed sloppily in this specific situation, I could have very easily justified skimping on quality. But I chose to be professional and did the best I could and refused to let his attitude manipulate me into acting like an amateur. I then some.

One day I found a call on my answering machine from a huge chain of hair-styling salons. The regional coordinator of their in-house agency had happened to sit next to this stereo shop proprietor at a fund raising dinner for a local politician. In the course of their shop talk at the buffet table, the lady from the hairstyling salons complimented my client on his advertising and insisted on being put in contact with me. She felt that I was performing as a true professional, and she now hires me for work that grosses a tidy annual sum. And I'm still slaving over the stereo shop's account for which I am paid a couple of hundred dollars a year.

What are we missing? You are in the best position to honestly evaluate weaknesses and try to correct them. Don't try to kid yourself that this doesn't apply to you, or worse, that you will be just fine even if you don't get organized. Because you really won't. You may stay in business for a while, and you might be having fun, but you'll be wasting so much. As long as you are already talented and motivated, why not take advantage of one hundred percent of your potential? It's already yours. You own it. It's not transferable. You can't give away your skill or passion as a gift. But you can sell it! ■

Editorial

One of the more difficult jobs for an editor is to try to discern what articles and features the reader wants. We do have knowledge of who you are: You are recording studio, broadcast audio, and sound contracting/reinforcement working engineers and technicians. Those of you in these categories are the backbone of our readership, starting with our very first issue (for some of you) in 1967.

Recently, we added a new category of readers: The smaller studio operator/engineer. This area of the pro-audio market is the fastest growing area in the industry, so it was logical that we cover (in our 2 to 8 trk section) material of special interest to that group. Imagine, then, our surprise to find that we are now getting letters from our traditional readers telling us how useful this new section and our directories are.

Conversely, the large readership group we inherited from the music industry technology magazine we once published, are telling us how useful they find the engineering articles we continue to feature in each issue.

What does this mean? We think it means just what it implies: The broad spectrum of recording studio operators, from the largest 24-plus tracks to a basement 8 tracker share a need to find out more about their craft. What are the new products and what do they do for me? This applies to several articles in this issue alone.

In **Personal Sound** in our 2 to 8trk Section, Ken Selcer describes how a musician becomes a studio owner, starting and remaining "small" by major studio standards.

At the other end of the spectrum, Murray Allen who runs Chicago's most sophisticated and successful major operation, describes his newest "toy" that will make his work better.

These are the range of articles to be found in every issue of **db, the Sound Engineering Magazine**. And for 1987 there will be more. More for <u>every</u> spectrum of the pro-audio engineer.

LZ

Murray R. Allen

Looking into the Future to Improve the Past

Universal Studios of Chicago, Illinois recenty acquired a Synclavier synthesizer for the studio operation. It would seem that it may be taking over the studio.

THIRTEEN YEARS AGO WHEN I recorded with Bill Chase I wished for a recording technology that would give the head room necessary to really capture that powerful brass sound. The following year I recorded Stan Kenton who insisted the drums be out in the open studio with all the brass and saxes *sans* baffles. I wished there was a recording technology that would allow one to accomplish this and still be able to control the presence on all the microphones.

While film mixing, I wished there was a recording technology that would afford me a more effective method of dealing with sound effects. In fact I wished there was some technology that would cut down on the quantity of six tape reels and records that always caused a great clutter in our mixing rooms.

When working with rock and roll I occasionally would record a bass player with a lovely sound and a terrible sense of rhythm. If only there was a recording technology that could make up for bad time and pitch. Sometimes I would be asked to slow a track down so the performer could play his part more accurately. Even with the best harmonizer, large variations in speed were impossible. If only there was a recording technique that was independent of speed without pitch variations.

When mixing in the domain of video sweetening why did I have to wait a few seconds until the machines locked up? why couldn't someone develop a technology that would cause an instantaneous lockup?

Well someone has!!!

WHAT HAS SOMEONE DONE?

A company called New England Digital has come up with a device called the Synclavier that solves all of the above problems.

To begin with it is a digital device. What this means is that the audio signal is converted into ones and zeros and stored in RAM or on Winchester disks much in the same way as business office computer systems store their bookkeeping information. The secret to success in this endeavor is the pureness of the analog to digital converter. We all know that sound as it passes through the air is in the analog state. When it is converted to ones and zeros it is important that the converter device adds as little coloration to the sound as possible. The converters used by Synclavier are the best I have heard in the marketplace. While performing A to B tests and in/out tests some of the best ears have had trouble telling the converted sound from the original.

Next, all the control functions are digital. This means that if you are at the end of a recording, you do not have to rewind to get back to the beginning.

GETTING THE BETTER RECORDING

How would I get a better recording of Bill Chase by using the Synclavier? Let me count the ways.

First because the Synclavier is a digital recorder it by definition has more headroom than any piece of analog tape could ever dream of attaining. When Bill punched out those high notes his peaks exceeded his RMS by 20 to 30 dB. If I recorded Bill at -5 VU 250 nw/m at 30 in/sec any peak over 20 dB would go down the drain. The result would be difficulty in determining any difference in a real hard punched note. It was these minute nuances that made Bill's performance so exciting. Another feature of the digital domain is the absence of wow and flutter. Bill liked to lay down several solo tracks and ping pong to get the best results. Even though we ran at 30 in/sec on the most tweeked-up decks, I still always felt a softness created by the minor wow and flutter more inherent on analog recordings. When we moved to another studio in another city to mix, the equipment we used was older and more abused and this problem became even more apparent. In digital, if it plays back, it will playback the same regardless of where you are and the usage of the equipment.

When recording Stan Kenton I had to tight mic the drums to keep the brass leakage to a minimum. This band played so loud that the drums did not leak into the horn mics. The problem was the reverse of normal. When one places cymbal and hi-hat mics too close to their sound source, they just don't sound natural. All the equalization and all the king's men could not restore these sounds to their natural state. By using the Synclavier I could sample Peter Erskine's cymbals and hi-hat by placing *mics* at an optimum distance to acoustically pick up the best sound. After recording the entire band *in situ*, I would replace the tightly-miked cymbals and hi-hat with more correctly recorded counterparts. This would also eliminate the sound of a brass section bouncing off the cymbals. Of all the types of leakage, this is the most undesirable.

SOUND EFFECTS

Historically, sound effects (sfx) for film have been handled in two ways: First, the sfx are pulled out of the studio library or recorded on location. Second, these sfx are transferred to 35 mm magnetic film and edited into the right position by a film editor. The second method is called Foley. Foley is the technique whereby a sfx specialist actually recreates

their sound effects while watching the picture. Because of this Foley stages have been very large. They have several pits filled with water, gravel, sand and other materials the sfx specialist might have to walk through, fall in or do what ever else is necessary to create their desired sfx.

EXIT THE FOLEY

By using the Synclavier, there is no need for a large Foley stage. To get footsteps walking through gravel, you sample a few footsteps walking through gravel. When you create the patch, the footsteps are spread all across the keyboard. This gives one the ability to change the pitch of the footsteps, even creating a dopler effect as they run by. This was impossible to do on the traditional Foley stages. The patch can also be made to reflect the speed the keyboard is struck. The harder the stroke, the louder the footsteps. If a scene is too fast, the entire picture can be slowed down to a speed comfortable for the foley artist to perform accurately. This was impossible on the traditional Foley stage. If the sync is off because of the reaction time of the Foley artist, with a twist of a knob it will be brought back in sync. If a few effects are slightly out of sync, the computer operator has only to type in the correct frame numbers for the erroneous events and we will be in sync again. The other day we had a bass player in the studio who could not play in tune; however, he had the perfect sound for the recording. We took sample of his sound and made a patch for the Synclavier. He then sat down at the keyboard and played his part (in tune) and then went back and moved any note he felt might be more effective if it hit slightly earlier or slightly later. When this process was completed, he felt it was too perfect and that a little controlled out-of-tune might not be a bad thing. He then worked on adding in only a few spots some faulty pitch and then smart correction. The final result was a perfectly human bass track with the best of all worlds.

We have other clients that prepare MIDI tracks in their office. They then pick sounds on the Synclavier to be triggered by their MIDI track. This takes a great deal of cost out of the creative process and opens the door of really high quality professional sounds for the beginning producer as well as the experienced composer. I feel this is the direction the home studio will take. The composer will create at home and then he/she will take their end result to a professional studio where the computer information will be processed into a glowing finished product.

The major studios love this concept. Their doors will be open and affordable to everyone and everyone will have access to the best in sound and experience. We call this the mothership concept. Who knows, the day might come where a musician might be able to phone his part into the studio.

The next wonderful thing about Synclavier is its ability to instantly lock up to picture. The moment it sees time code it is in lock. This means you can run at any speed and be locked up. How about one frame a second for checking really tight cues?

I am sorry I don't get a chance to mix anymore. With the wonderful tools that are available today and especially the Synclavier, my job sure would have been a great deal easier. I wonder: What would I look like without gray hair? ■

George Blood

My Two Weeks on Studer Row

A small console and the Philadelphia Orchestra in its summer concert series–a revelation!

IT WAS ANOTHER DAY OF irregular weather in Saratoga Springs, New York. Frost warnings at night, record high temperatures and high humidity during the day. But for all I knew, it was just another Thursday. Another 9 AM call, another 11 PM strike. More horse stories. More mic placement complaints. I had no reason to suspect that this day–or even the next few weeks–would be different.

It was the same routine 150 times a year. Set up before the musicians arrive, strike when they're gone. The Orchestra is a good band, mind you, but it has 100-plus personalities and just as many priorities. It does make a nice change from the city, but it means a grueling schedule: one morning rehearsal, one evening concert. There are no second chances, no "take it from another night," no great experimentation.

George Blood records and edits concerts of the Philadelphia Orchestra for international radio syndication by WFMT, Chicago.

THE MIXER COMES

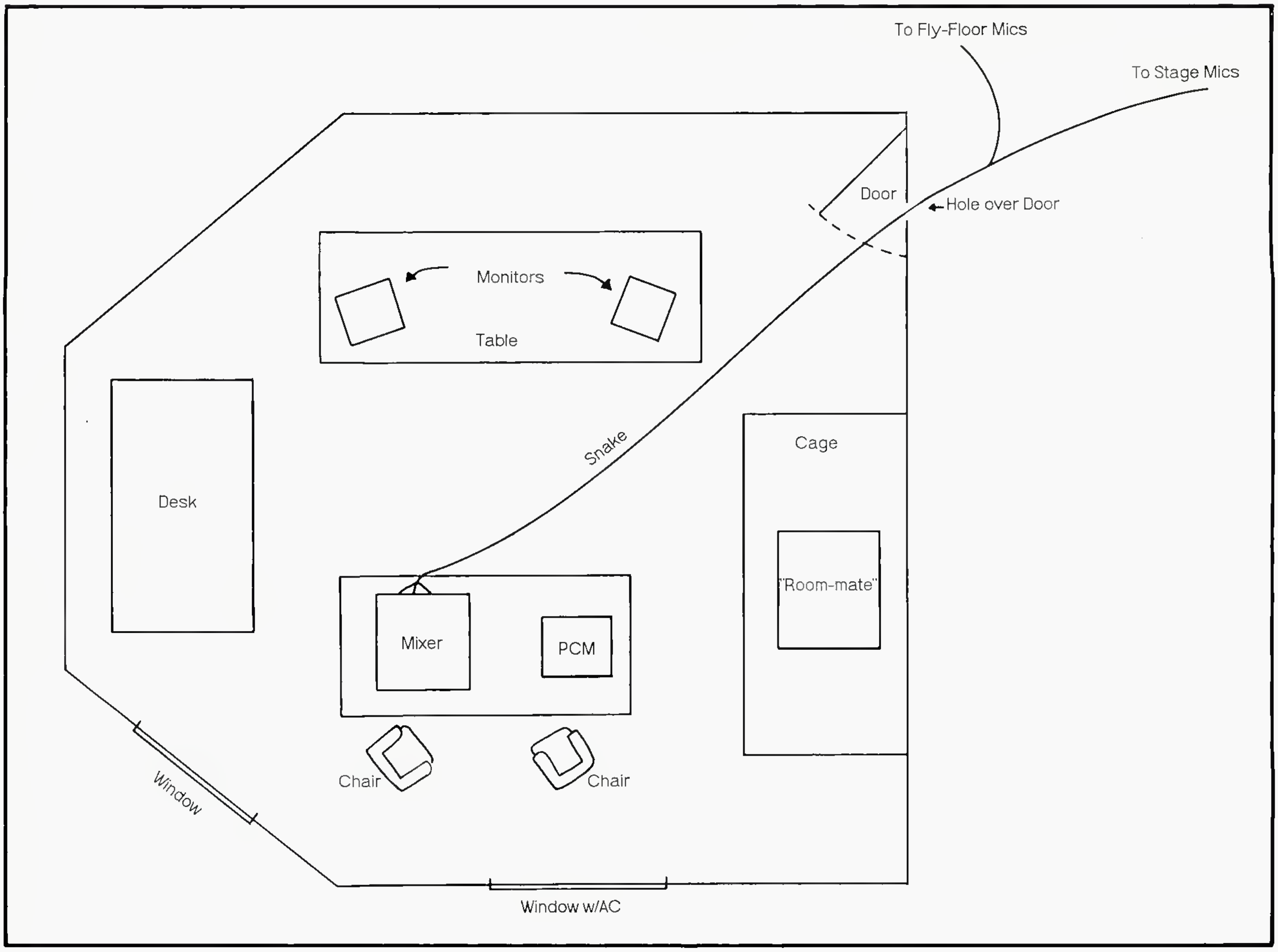
Since it was about lunchtime, the rehearsal was almost over. Big Nick, one of the stage hands, came in with a handtruck and a big box. He turned to me and demurely asked, "Is this for you?"

I couldn't believe it. Brian had finally done it. After weeks of telephone tag, confusion, and exchanged favors, I finally got a Studer 961 portable mixer for two weeks.

It came directly from its previous demo with no time for a quick quality check in Nashville. It would have to show me what it had. Was it really portable? Could it really be shipped cross-country through God only knows what kind of handling? Would it really do what they say it would? Well, I couldn't wait to find out.

We have both heard a lot about the Studer 961 and yet there was something very different about it. Perhaps its size or its sexy Studer gray coat. Perhaps its reputation–although limited.

The curiosity would have to wait until tomorrow,



The master control room.

for the levels for today's show was already set and there was no time to make any changes.

INSTALLING THE BOARD

Friday, 9 AM. Larry and I returned to the amphitheater to switch mixers. It's really a simple chore which involves some patching and phase switching. We had to switch from 8 x 2 with pin 3 high, to 10 x 2 with pin 2 high. The hardest part of the switch was adjusting the trim pots. But that was just a mere bag of shells compared to some of the other problems. During the night, the huge temperature swings and humidity caused condensation on the microphones. We had to take the mics down and leave them on top of the power supplies to dry. This is the last time I whine about a hot power supply.

With all these details out of the way, and rehearsal about to begin, Larry and I could get down to what we had been waiting for, listening. As I marveled at all the knobs, Larry commented about the sound. Does it actually sound better? Or was it just that Studer gray again? It was all going to PCM, with all nine inputs up at zero. It wasn't adding any more noise than the dither was, and the mixer didn't seem to mind the room.

THE ROOM

Let me explain the room. The theatre was a five-sided cinderblock with a high ceiling and flourescent lights. It was one of the few rooms in the ampitheatre with an air conditioner. Unfortunately, you can't run the air conditioner during classical shows because it makes too much noise.

Because of the air conditioner, we had to bring in a Kliegl with ninety-six 2.4 kilowatt dimmers that made more noise than their cooling fans. But the 961 mixer just ingored them.

I have seen a good number of mixing boards in my day (some that were better described as consoles) and I know what I like and don't like. Regardless of my personal feelings, I had this 961 to play with. So what did it have to offer?

WHAT DOES IT HAVE?

For starters, the input selector contained the choice of sensitivities for mic, line (with a concentric trimpot), and a choice of generator and off. Phantom power, switch selectable high pass filter with cut-off frequency of 75 Hz, 12 dB/octave, and phase reverse. The phase reverse showed an improvement over the 169 in that it was a hard wired phase flop. But here, as in the rest of the 961, it's all FET switching. The equalizer section has high and low shelving, +/-15 dB, a mid-0range with a range of 100 Hz to 7 kHz, and an on/off. Again, all pots were the concentric type.

The two sends, either pre- or post-fader, were front panel selectable with a pop-up pot. Due to the flexibility of modular design, the mixer is available with

one to four output channels. Moving down the mixer, we find the mute switch with indicator, prefader solo, and of course linear faders.

As usual, I had many visitors during my two weeks with the 961, and without exception, no matter what they thought of the rest of the board, they all liked the faders. They were more than mere conductive plastic with a range from +10 to -60 and beyond. They had a special feel which was not quite as smooth as glass, but rather as smooth as a baby's bottom. No "spots," no overshoot, no fade (no pun intended). They went where you put them, and stayed there. They moved only when you wanted them to, and only to where you wanted, but no more.

After drooling over these faders, it occurred to me that not only the faders, but everything from the switches to the meters and power supplies, worked exactly as the textbooks say they should. When at center, the pan pots panned center. Dead center. If I put percussion on one input, and oboes on the next, the first channel would be percussion, and the second would be oboes. Nothing else, no bleed. If I pushed the mute switch, it was gone. Completely gone. Everything worked perfectly. Well, almost everything.

SOMETHING WAS MISSING

Larry was the first to notice that something was missing. Something wasn't working. Then it hit us–there were no lights! No mute indicators, no overload lights, no power switch lights. After a quick check of the manual–which provided no clues–we found a phone and called Nashville. Doug, the service technician, was in a meeting, so we left a message.

An hour later Big Nick came running over to tell us that Doug was on the phone. "What seems to be the problem?," he innocently asked. "We ain't got no lights," I screamed. Once he explained what the most likely problem was, we were off to fix the problem. A blown fuse it seems. How did the fuse blow? We didn't remember any lights when we first plugged the unit in.

Once we found the power supply tucked neatly under the small table at the end of the mixer, the rest was easy. Turn two screws and the front panel is removed exposing the optional batteries and the main fuse. Turning a few more screws releases the entire power supply. Once inside, we found the location of the blown 6 volt fuse. After a quick trip to the local Radio Shack (they are everywhere) we had lights. Pretty red and orange LEDs.

Back to work. The master output channels, which are just about as busy as the inputs, have the same "something to write home about faders" and mute switch. The same modules carry the high level inputs for overdubs, reverb return, etc., and has a rotary level pot, the same aux sends, PFL, and channel select/pan pot arrangements on larger knobs. And last, but far from least is the on-board limiter.

USING THE LIMITER

It's an innovative idea–a field mixer with a limiter on board. It provides adjustable ratio, 1:1 to 1: infinity with variable release time, gain before compressor, intra-channel linkage, on/off, and insert. Insert? To find out what this was, I had to make another trip to the manual.

This is a great idea. The Studer gives the limiter a patch point (on the rear panel) and with the insert depressed, you can use the limiter on an individual input channel. The indicator lights for the limiter were odd, though I could never tell if it was working. There was no pumping, choked transients, or "uh oh, there goes the limiter." The literature indicates this is PDM, Pulse Duration Modulation, but since I am uneducated, all I had to do was listen to the thing. The limiter actually listens to the music. "This is loud, this is soft, this is not so loud, this is fine." Period. It would take a far better man (or worse depending on your perspective) to do harm with this limiter.

The rest of the mixer is filled with options that are nice to have. A "CR Monitor" section is used to chose PFL, AUX 1 or 2, and a couple of user selectables. "PFL to Monitor" selects output instead of on board PFL speaker for PFL'ing. A "Mono" switch, and a level control with concentric balance is also included. It also has a very elaborate "everything to everybody" headphone section. You can chose to hear the monitor output, PFL, or a combination, one in each ear. There are two jacks, one that provides just the output, and the other which disables the PFL speaker on the meter bridge. And of course there is a level control here as well.

On the last module are the talkback facilities, the "To Studio" and "Slate" select, AUX send master levels with their own talkback selects, TB mic, the generator with on/off plus six frequencies, and on/off for the Littlite (included) which attaches via a BNC connector.

Then come the meters. They are choice of VU or PPM with front panel speed selection for each output. There are AUX level meters and gain reduction meters for the limiter, and as an option, you can get a correlation meter for checking phase.

After using it for a few days, Larry and I spent some time trying to decide what was worth having on this board and what was it that made it worth the price, if anything. Although Larry had his doubts about the eq, I think it fits in with the rest of the board–it suffices. Not that the 196 is just "good enough," but on a board of this size–less than the size of a small suitcase (but weighing 55 lbs.)–you can't waste space. There is only so much room inside, but there is plenty of room on the front for switches and the like. Yet the layout and construction are clean, the components all look good, and everything fits. And it honestly looks serviceable. With the 4-channel version, you can easily have the board set up for broadcast use–a stereo program and stereo audition.

In short, the Studer 961 fills a specialized market, and does it very well. But it's not without its competitors. The Neve 542, the old MCI JH-800 and the Sonosax mixers are all viable competitors. Your choice not only depends on product loyalty, but also a matter of your use. I will leave you to investigate which does what for yourself.

But instead of going on and on, I refer all interested to your friendly Studer rep to ask for PI 14/85 (that's for product information). That's how I got started.

And alas, all good things must come to an end, and I eventually found myself driving to New York City to return the 961. But it's a trip I hope to be making again in the very near future. ■

Jon Gaines

Construction Project: A Low Distortion Sinewave Oscillator

All you ever wanted to know about why and how to build this essential audio tool.

Completed sine wave oscillator.

An audio oscillator is an invaluable tool for any studio or audio facility. It may be used as a trouble-shooting aid, in new product design, and for such mundane chores as putting alignment tones on master tapes. If you've looked for oscillators in standard electronics catalogs, however, you know that much of what is available is either more than you wanted or unsatisfactory in some way. For example, a professional signal generator designed for lab use may cover a frequency range of 0.1 Hz to 50 Gigahertz in half cycle steps with eight multipliers and ten attenuators when all you really wanted was something in the audio range. At the other extreme are the inexpensive function generators based on IC oscillators that can be trimmed for less than 1% distortion, making them unsuitable for analyzing filters and equalizers and useless for performing distortion measurements.

The subject of this article is a low cost, easy to build, low distortion sine wave oscillator with switch selectable at 20, 50, 100, 200,and 500 Hz, 1, 2, 10, and 20 kHz. Typical distortion is <0.01 at mid-band frequencies. Output level is adjustable up to a maximum of +20 dBV, and will drive either low or high impedance loads.

THE CIRCUIT

This project is based on a classic oscillator circuit, the Wein Bridge. It employs an op-amp with both positive and negative feedback elements in the

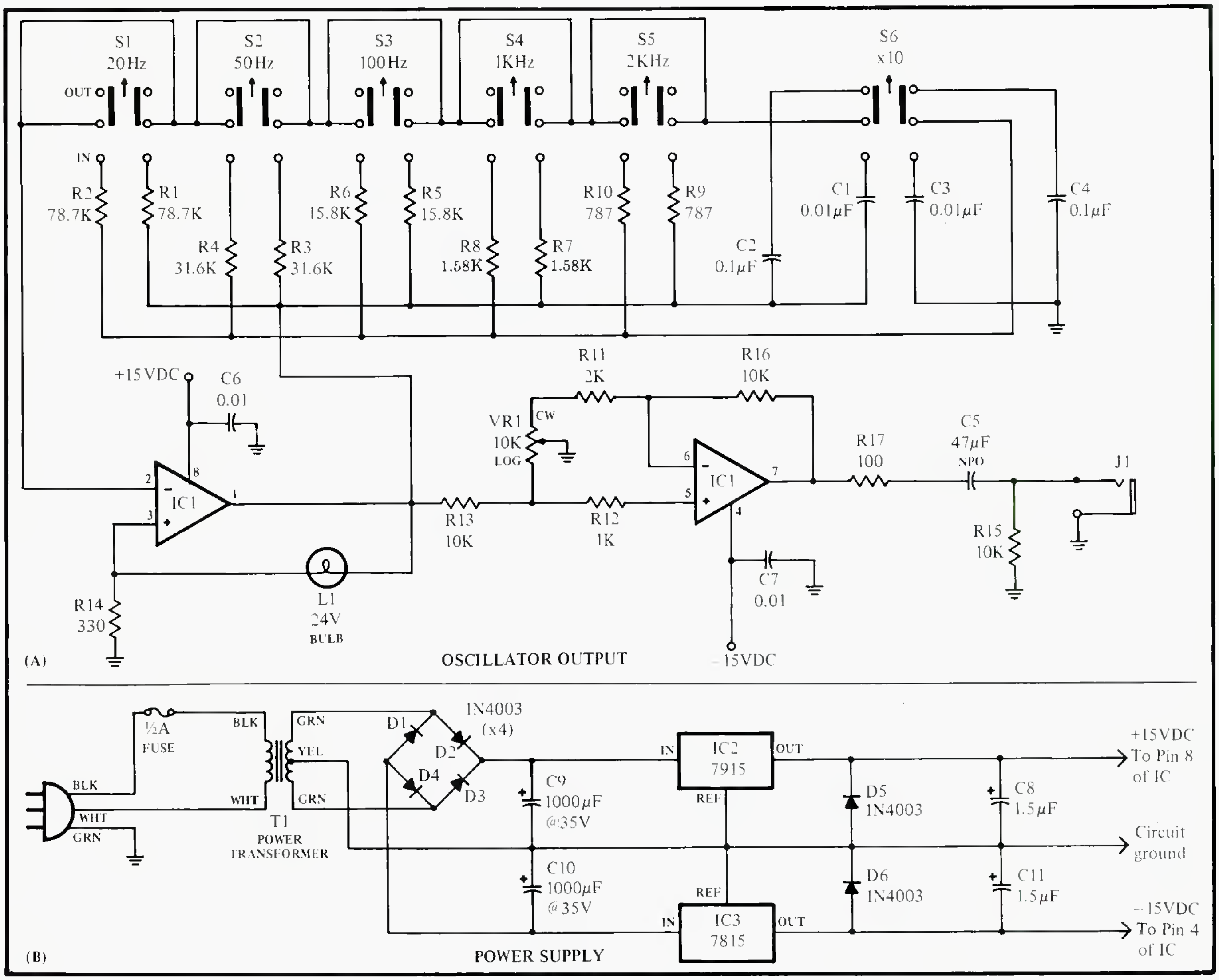


Figure 1. The schematic diagram for the oscillator circuit (A), and power supply circuit (B).

negative feedback loop, with positive feedback being maintained and controlled by an automatic gain control in the form of a light bulb, L1. This light bulb doesn't actually light up, but its resistance varies in proportion to the current through it, providing a neat variable resistance element.

The circuit requires amazingly few parts to do what it does so well, and this makes a good kit for anyone who is interested in building their own test equipment.

For convenience, this project is self-powered with an integral +/-15V power supply. If you already have such a supply you may use it instead, although it's nice to build this as a self-contained and portable unit so that you can carry it around with you. You can also power it with four 9V batteries (+/-18V) for portable use.

BUILDING THE OS2 OSCILLATOR

You can build the oscillator either on a piece of *perfboard* or on a printed circuit board. Full size artwork for the PC board is shown in *Figure 2* and you may etch and drill your own or purchase one prefabricated (see parts list). In either case, assembly is as easy as plugging in the appropriate components by referring to *Figure 3*, the Component Assembly Guide. Pay particular attention to the polarity of diodes, electrolytic capacitors, and the IC. It's a good idea to heat-sink the voltage regulators.

Solder all components carefully with a medium heat pencil type soldering iron and resin core solder, them trim all excess lead lengths. Check your work for correct component placement, solder bridges, and cold solder joints.

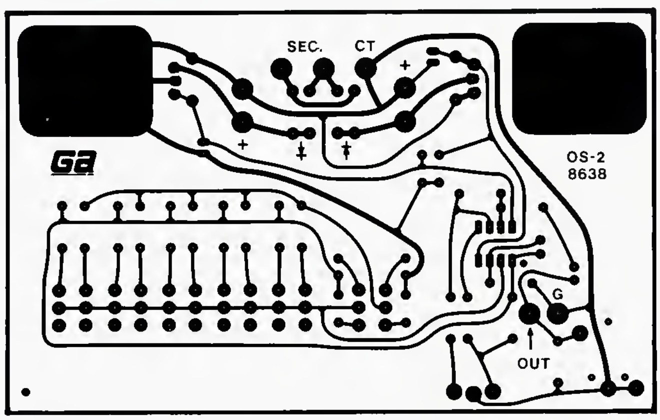


Figure 2. The foil side of the printed circuit board.

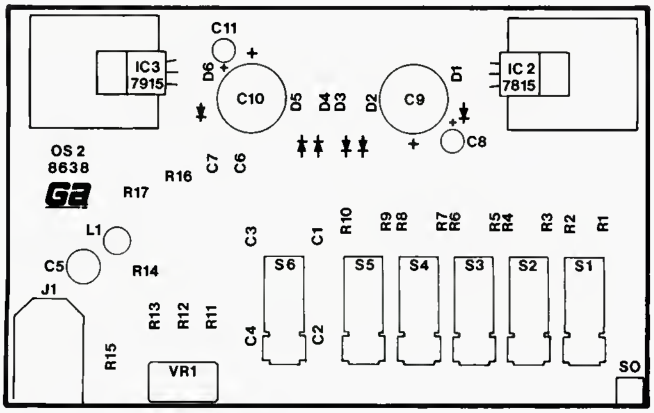


Figure 3. A component assembly guide showing component placement on the PC board.

EXTERNAL WIRING

Since all components in the kit plug right into the PC board, the only wiring necessary is for the AC side of the power supply. Start with a grounded (3-wire) line cord. Securely attach its green wire (ground) to the chassis of the oscillator. The white (neutral) wire in the line chord attaches directly to one wire (white) of the power transformer primary using a wire nut or crimp connector. The remaining black wire of the line cord (hot) attaches first to the fuse holder and then proceeds to the remaining primary wire (black) of the power transformer. Make sure that all exposed AC connections get covered with shrink tubing or electrical tape. If you wish to use a power switch, connect it between the fuse holder and the primary of the transformer.

The transformer secondary center tap (yellow) connects to the circuit board at the point marked CT, and the remaining two green wires from the power transformer secondary connect to the two pads marked SEC. That completes all the essential wiring. Again, trim any excess lead lengths and check your work.

There are still two unused holes in the PC board near the output jack, marked G and OUT. These pads may be used to connect a parallel output connector on the back panel of the unit such as an RCA or BNC connector. Or, you might want to install the oscillator in your rack or hard wire it into a mixing console while leaving the front panel jack free for other uses.

A FEW WORDS ABOUT SAFETY

Since this project has an internal power supply and you will be working with the 115V AC line, you must use common sense and standard safety practices. In particular, never work on the unit when it is plugged in, and never plug it in until you're sure that all connections are correct and that no AC connections are exposed or in contact with the chassis. Always fuse the AC line and NEVER substitute a higher value fuse if the correct one blows. Use a grounded 3-wire AC cord and never cut off or defeat the

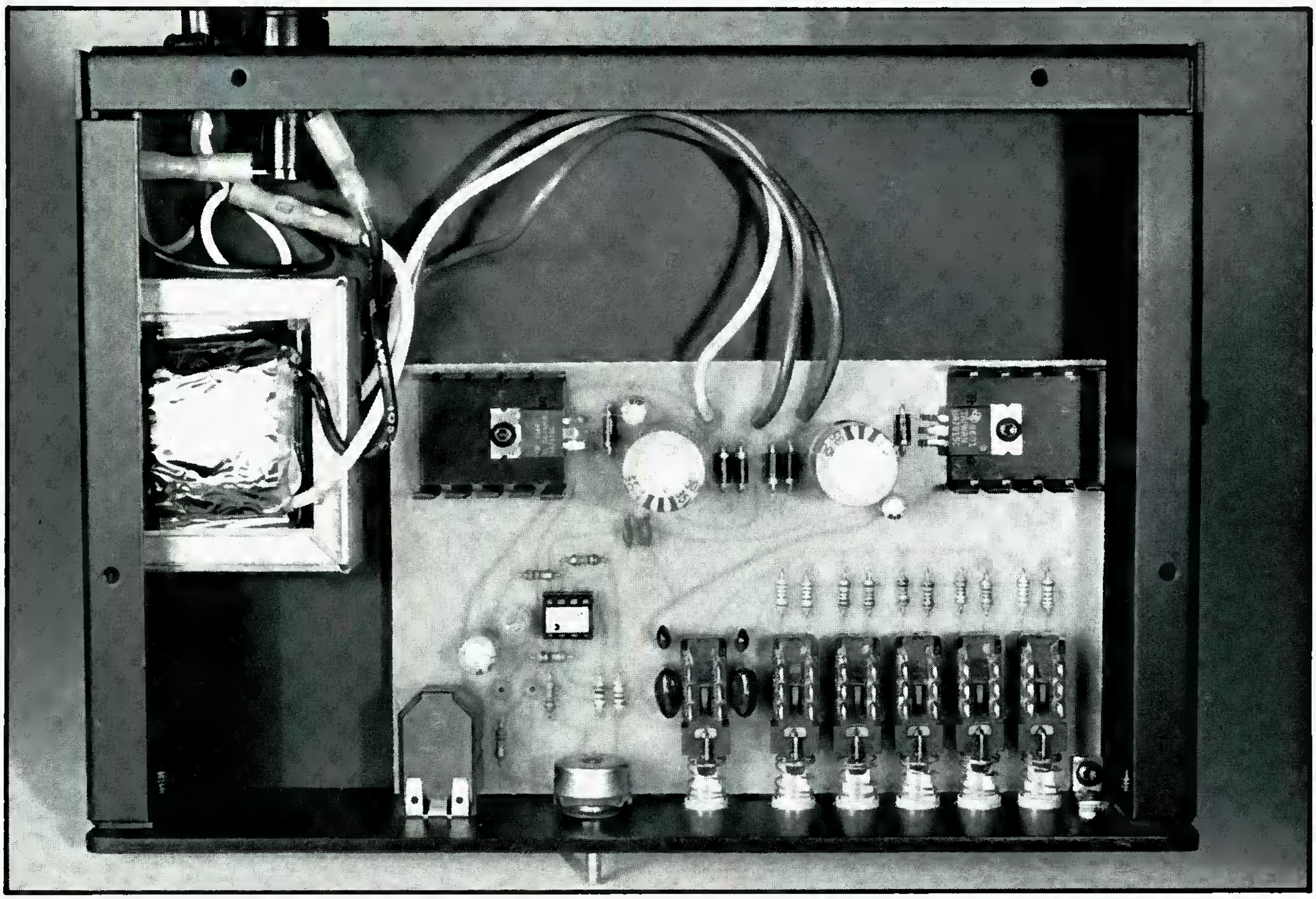


Figure 4. The completed oscillator with bottom cover removed.

ground pin. Most important, if you have any doubt about your ability to handle the project safely, stop and get help from a qualified technician.

TESTING THE COMPLETED PROJECT

After you've soldered and trimmed all connections, double check your work for correct component placement and orientation. Look again for shorts and cold solder joints. When you're satisfied that all is well, plug in the unit and check the power supply with a DC voltmeter. You should see +15V DC on Pin 8 of the IC socket and -15V on Pin 4. If not, go back and look for power supply errors. If you immediately blow a fuse when plugging the unit in, look for errors in the AC wiring or incorrectly installed diodes or electrolytic capacitors.

With a correctly functioning power supply, the oscillator should come right on and work as soon as you've plugged in the IC and pushed one of the frequency select buttons. Hook it up to your monitor system or even plug in a pair of medium impedance headphones at its output jack. Choose a mid-frequency such as 1K Hz and gradually turn up the level control. If you have an oscilloscope available, you'll also want to observe the output and should expect to see a perfect sine wave.

Depending on the specific light bulb you're using, you may need to adjust the value of R14 during this initial setup check. With the correct value resistor in place, the oscillator will produce its maximum output level without clipping. If the output is clipped or the oscillator won't produce the specified +20 dBV output level when you turn the level control all the way up, try some different values for R14 or even install a 1 K-ohm trimmer to find the exact value.

USING THE OS2 OSCILLATOR

Operation of this kit is pretty obvious; just push a button to select a frequency and adjust the output level as needed. The x10 switch, of course, multiplies any chosen frequency by a factor of 10, so 2 kHz becomes 20 kHz, etc. Note that although the selector buttons would seem to indicate a total of nine available frequencies, you can produce additional intermediate frequencies by depressing two or three buttons simultaneously. For example, pushing both the 50 and 100 Hz buttons down will produce an output frequency of 150 Hz.Pushing the 1 kHz and 2 kHz buttons will produce 3 kHz, and so on.

You'll notice that the output level will *bounce around* a bit when changing selected frequencies, but it settles quickly and remains stable until changed.

The accuracy of the chosen frequency is dependent on the tolerance of resistors 1-10 and capacitors 1-4. 1% resistors and 2% tolerance polyester film capacitors are recommended although other parts may be substituted with a resulting decrease in accuracy. Also, note that you can select other frequency centers than the ones I've chosen for this project by changing any pair of resistors located right behind the selector switch; the higher the value of the resistors, the lower the output frequency.

PARTS LIST

RESISTORS

R1, 2	78.7k ohm, 1% metal film
R3, 4	31.6k ohm, " "
R5, 6	15.8k ohm, " "
R7, 8	1.58k ohm, " "
R9, 10	787 ohm, " "
R11	2k ohm, 5% carbon file
R12	1k ohm, " "
R13, 15, 16	10k ohm, " " (see text)
R14	330 ohm, " "
VR1	10k ohm Log taper potentiometer

Capacitors (all values are in Microfarads)

C1, 3	0.01 Polyester Film, 2% tolerance
C2, 4	0.1
C5	47 @10V, Non-polarized electrolytic
C6, 7	0.01 ceramic disks
C8, 11	1 or 1.5 @50V, electrolytic
C9, 10	1,000 @ 35V

Other Parts

D1-6	1N4003 Silicon Rectifier Diodes
*ICI	Dual Op-Amp, TL072
IC2	Positive Voltage Regulator, LM7815
IC3	Negative " , LM7915
S1-6	Shadow LT series switches, DPDT
T1	Power Transformer, 45V AC Secondary, CT
J1	Phone Jack
F1	1/2 Amp fuse, fast blow
L1	24V light bulb

OS2 Printed Circuit Board
Knobs for pot and switches
Line cord, grounded
Wire nut or crimp connector
Fuse holder
Strain relief
Heat sinks
Chassis
Misc. hardware
Standoff (secures circuit board to front panel)

KIT AVAILABILITY

The following items are available from Gaines Audio, PO Box 17888, Rochester, NY 14617 (716) 266-0780:

0S2 Printed Circuit Board: $11.95, postage paid

Complete kit as shown, including chassis: $74.50

Assembled and tested 0S2: $99.50

Please add $2.25 for shipping per item except on PCB. New York State residents please add 7% sales tax.

All orders are shipped promptly and come with a 30-day return privilege if not satisfied for any reason. VISA, MC, Money Orders, and Certified Checks accepted. Personal checks must clear prior to shipment. ■

Bruce Bartlett

Measurements and Perceptions

Sometimes what you think will record or sound one way, will turn out an entirely different way. This article tells why this happens and what there is to do about it.

WHY DOES A RECORDED BASS GUITAR lose its clarity when other instruments are added to the mix? How distortion-free does a sound system have to be to sound clean? How audible are deviations from flat frequency response?

These questions are addressed by a science called *psycho-acoustics*, which deals with the way we hear things. It studies the relation between objective measurements and subjective perceptions. That is, it relates what we measure to what we hear.

The findings of psycho-acoustics have far-reaching implications in recording, as we shall see. We'll look at several different kinds of audio measurements, relate them to how we perceive them, and point out their importance in audio.

FREQUENCY RANGE

Let's start by asking, "What frequency range is needed for high-fidelity reproduction? If we want to cover the full range of human hearing, what is that range?"

Children can hear frequencies from 20 Hz to 20,000 Hz. Most adults can hear up to 15kHz, although they lose high-frequency sensitivity with age. So, if we want to reproduce all the frequencies anyone can hear, we need a system frequency range from 20 Hz to 20,000 Hz. The range of 40 to 15,000 Hz covers most musical sound sources.

The required frequency range for a microphone depends on the sound source. An orchestra produces frequencies from 40 Hz to 15,000 Hz; so the recording microphones must reproduce at least this range. On the other hand, if you're recording an instrument that produces frequencies from 80 Hz to 10,000 Hz, then a microphone response from 80 to 10,000 Hz is adequate.

FREQUENCY RESPONSE OF THE EAR

The frequency response of the ear is not flat. We are most sensitive to frequencies around 4kHz, and less sensitive to high and low frequencies. Low frequencies must be much more intense than mid frequencies to sound equally loud.

This relationship is shown in *Figure 1*, the Robinson-Dadson equal loudness curves. They were derived as follows: Listeners, seated in an anechoic chamber, listened with both ears to tones of different frequencies and intensities. They compared the loudness of each tone to that of a 1000-Hz tone at a fixed level. The equal-loudness contours show the sound pressure levels of pure tones of various frequencies that were judged to be as loud as the 1000-Hz reference tone. The experiment was repeated with different reference-tone levels, producing a family of equal-loudness contours.

As the contours show, low frequencies require a much higher SPL than mid-frequencies to be equally loud. In other words, the ear is less sensitive to low frequencies than it is to mid frequencies.

We're used to hearing sounds naturally with this frequency response, so there's no need to boost the lows and highs in a recording just to compensate for the ear's non-flat response.

Note that the curves are not parallel at low frequencies; the curves tend to flatten out at high SPLs. So, at low volumes, we hear less bass than we do at high volumes. The perceived tonal balance of a musical program changes with playback level.

This fact has important consequences in mixing. Suppose you do a mix while monitoring extremely loudly. If the end listener plays this mix at a moderate level (say 85 dB SPL) he or she will hear less bass in the mix than you did while mixing. Recordings meant to be played at 85 dB SPL (a typical home listening level) should be monitored and mixed at that level. If you mix at high SPLs, you'll fool yourself into thinking you have plenty of bass in the mix–but the home listener may not hear it.

To capture an accurate tonal balance, classical

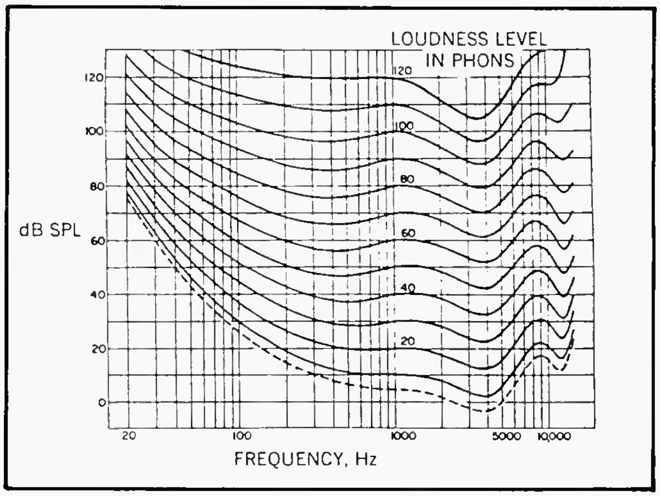


Figure 1. Robinson-Dadson equal-loudness contours.

music is generally recorded with flat-response mics and little or no EQ. If home listeners want to hear a classical recording with the same tonal balance as the original performance, they must play the recording as loud as the original performance. Lower playback levels will result in weak bass.

The ear loses high-frequency response (mostly around 4 kHz) for awhile after being exposed to loud sound. This is called *temporary threshold shift.* You may have experienced this phenomenon after attending a loud rock concert–your ears ring and you can't hear high frequencies for several hours after the concert. Later, hearing returns to normal unless permanent damage has been done.

If you monitor recordings loudly for long periods, you may find yourself gradually turning up the high-frequency EQ as your hearing loses high end. It helps to monitor at moderate levels or take frequent breaks to keep your hearing fresh.

SPECTRUM AND TIMBRE

The *spectrum* of an instrument is its output vs. frequency–the number and relative levels of its fundamentals and harmonics. The perception of the spectrum is called *timbre* or *tone quality*.

Timbre also is determined by the transient character of the instrument: the envelope of attack, decay, sustain, and release. Different instruments have different rates of buildup and decay for the various harmonics.

To prove how important transients are in defining the sound of an instrument, try this: Edit out the attacks from a series of recorded piano notes. The piano will start to sound like an organ.

FREQUENCY RESPONSE

The *frequency response* of an audio device is its output level vs. frequency, when driven with a signal that is constant in level with frequency. For example, if you drive an amplifier with a sine-wave sweep that is the same voltage at all frequencies, the output voltage of the amplifier vs. frequency is its frequency response.

A flat frequency response–uniform amplitude with frequency–is a requirement for natural or lifelike reproduction. Here's why: A musical instrument produces a certain spectrum at the listener's ears. This spectrum identifies the instrument. To reproduce that same spectrum over a loudspeaker, the frequency response of the recording microphone and playback speaker must be flat. Then the fundamentals and harmonics will be reproduced in the same proportion as the live instrument. Since the recorded spectrum matches the live spectrum, the reproduced tonal balance is natural.

A microphone with a flat, extended frequency response, placed properly near an instrument, tends to provide natural reproduction of the instrument's tonal balance.

Deviations from flat response affect the reproduced tone quality. If you turn up the high frequencies with an equalizer, the harmonics are boosted relative to the fundamental. The result is a trebly, bright, crisp sound. If you boost low-frequency EQ, the fundamental is emphasized, and the sound becomes bassy and boomy.

Each frequency band in the audible spectrum, when boosted, provides a unique coloration. For example, a boost around 1kHz on speech sounds nasal, at 3kHz sounds metallic, at 5 kHz adds presence, etc. It pays to experiment with a graphic equalizer to hear the effects of boosts and cuts (response peaks and dips) at various frequencies.

How much peak or dip is audible? Here are the findings of one study:

With peaks and dips that are narrow (delta-f/f =0.35), 3-dB peaks and 10-dB dips are just audible on music. Peaks are much more audible than dips. Broad peaks are easier to hear than narrow peaks; we respond to the area under the curve. One-octave-wide dips, 5 dB deep, are just audible [1].

Some listeners can reliably detect broadband differences in frequency response of 0.2 dB. So, in tests comparing two audio components, their frequency responses must be matched to 0.1 dB [2].

It's generally thought that frequency-response aberrations narrower than 1/3 octave are inaudible [3]. However, Queen reports that peaks and dips 1/10 octave wide can be heard in the region of 800 Hz to 5000 Hz [4]. Toole also notes that response curves with resolution narrower than 1/3 octave are needed to correlate well with audible effects [5].

Kates reports that, for a signal added to its delayed repetition, the just-detectable peak/dip ratio is 1.9 dB [6].

Clark has made several findings related to the audible effects of frequency-response errors in loudspeakers [7]:

*Response notches are annoying if not filled in by sound reflections from room surfaces.

*Response notches or dips are almost inaudible if the notches are filled in by reflections within 10 msec. Thus, a speaker measurement correlating with what we hear should include not only the direct sound of the speaker, but at least 10 msec of room reflections as well.

*1/3-octave-wide peaks less than 1 dB from 500 Hz to 3kHz are inaudible. 1/3-octave-wide peaks less than 3 dB above 10 kHz or below 150 Hz are inaudible.

**Binaural echo suppression* is the ability of the ear to reject echoes coming from directions other than the sound source. Because of binaural echo suppression, wall reflections are hardly audible so they have little effect on perceived frequency response. Floor or console reflections are more audible because they come from approximately the same direction as the source.

Toole has further comments relating speaker response measurements to listening tests: "Experienced listeners with normal hearing (gave the highest fidelity ratings to) loud-speakers with wide bandwidth, flat and smooth amplitude response, and uniformly wide dispersion." The amplitude-response (frequency-response) measurement should be made anechoically at 2 m or more, with resolution exceeding 1/10 octave to show resonances. Measurements should be made on and off axis in the front hemisphere. Off-axis measurements should be spatially averaged to remove the visual clutter of acoustical interference effects [8].

Kates presented another criterion for loudspeaker evaluation in rooms. Stated very simply, reflections up to 70 msec are included in the frequency response measurement, which is smoothed by critical-band filtering. Kates found that the perceived response above about 2 kHz is independent of room reflections, but reflection effects between 100 Hz and 1000 Hz (usually caused by floor reflections) are clearly audible [9].

LOUDNESS AND PITCH

Loudness is the subjective correlate of Sound Pressure Level (SPL). A 6-to-10 dB increase in SPL is judged by most listeners to be twice as loud [10]. A 1-dB change in SPL is generally regarded as the smallest change we can hear, although the just-detectable difference varies from 0.1 to 5 dB, depending on frequency and bandwidth.

To make a sound twice as loud (an increase of 6 to 10 dB SPL), amplifier power output must be increased 4 to 10 times. Doubling the power increases loudness 3 dB.

Pitch is the subjective correlate of frequency. The higher the frequency, the higher the pitch. Doubling frequency raises pitch one octave.

High frequencies require much more frequency change for a given pitch change than do low frequencies. Doubling the frequency from 40 Hz to 80 Hz (a 40 Hz increase) raises the pitch one octave; so does doubling the frequency from 5 kHz to 10 kHz (a 5000 Hz increase). In general, the ear's sense of pitch varies logarithmically with frequency. That's why frequency response is usually plotted on a log-frequency scale. Such a scale has equal pitch change per octave, or equal pitch change for equal horizontal spacing.

For low-frequency tones, the pitch decreases as the intensity increases. The opposite is true for high-frequency tones. This fact affects the tuning of the bass guitar–the tuning may be off if done while monitoring very loudly over headphones.

MASKING

Masking is the inability to hear a given sound in the presence of another. For example, if you're talking in a car while a train goes by, the train masks the speech. You can't hear the speech during the loud train noise unless you yell.

Masking occurs during mixdowns. For example, a bass guitar may sound very clear when heard alone, but loses its clarity when other tracks are mixed in. That's because the other instruments mask the harmonics of the bass, dulling its sound.

Masking of one frequency by another is greatest when the two frequencies are close, and diminishes as the frequencies become farther apart [11]. For this reason, avoid having two similar-range instruments play together in a mix' otherwise they will mask each other and blend poorly. If two similar-range instruments are in a mix (say, piano and rhythm guitar), you can help keep their sounds distinct by equalizing them at different frequencies so that less masking occurs.

The louder the masking tone or noise, the greater the masking, and the more frequencies are masked. Low-pitched tones can mask high-pitched tones, but high-frequency tones have almost no masking effect on low tones [12].

DISTORTION

Distortion is the presence of frequencies at the output of a device that were not present in the input signal. *Harmonic distortion* has distortion products that are harmonics of the fundamental frequency. The first harmonic is the fundamental frequency; the second harmonic is twice the fundamental frequency; the third harmonic is three times the fundamental frequency, and so on. For example, even-order harmonic distortion of a 100-Hz signal appears at 200 Hz, 400 Hz, 600 Hz, etc.

Certain kinds of distortion have a gritty, grainy sound. To hear it, simply record at too-high a level and listen to the distorted playback. The distortion you're hearing is mainly odd-order harmonic. Distortion on voice sibilants, cymbals, or high soprano notes gives a mid-frequency *thickening* or *rushing* sound.

High-order harmonic distortion (from transistor amps or digital-recorder clipping) is more annoying or edgy-sounding than low-order harmonic distortion (tube-amp clipping, analog-recorder overload, or loudspeaker limiting)[13]. Even order distortion (associated with tube amplifiers) is said to give a euphonic, *open*, *choral* sound; while odd-order distortion associated with early transistor amps sounds more annoying, *closed*, or *covered*.

How great does harmonic distortion have to be before we can hear it? For low-order harmonic distortion on music, 1% is the minimum amount detectable by trained ears [14],[15],[16]. The same figure applies to crossover distortion on music. According to Russel & Fryer, 3% (1%-5%) is the just-detectable level [17]. Mark Davis reports that 2nd and 3rd-order harmonic distortion below 2% to 3% is inaudible on music, while 5th-order or higher distortion becomes audible around 0.1% to 0.2% [18]. Clark found that 3% even-order harmonic distortion is just audible on music [19].

It's much harder to hear distortion on music than on a sine wave, because the harmonics in the music mask the distortion. Sine-wave distortion of 0.1% is just audible [20].

Intermodulation distortion has components at the sum and difference of two frequencies. Difference-frequency distortion is below the frequency of the musical signal and, thus, is less likely to be masked than harmonic distortion.

Figure 2 shows the audibility of different types of distortion for music, speech, and sine waves.

Here are typical total harmonic distortion (THD) specfication for various components [21]:

Amplifiers: Less than 0.001% at low-to-mid frequencies, increasing to 0.05% at 20 kHz at rated power (Crown spec). Note: Harmonic distortion of frequencies above 10 kHz is inaudible because the distortion components are above 20kHz.

Phono cartridges: 6-8% (Shure spec). This figure is highly variable, depending on record-cutting level and condition of stylus and record. Distortion increases by a factor of 2 from the outside diameter of a record to the inside.

Open-reel analog tape recorders: 3% at maximum recording level (mostly odd order); 1% or less at 0 VU.

Cassette recorders: 4-6% at maximum recording level.

Tuners: 0.05% (Crown spec).

Monitor speakers: 1-2% at low frequencies, 0.3% at mid-frequencies at 80 dB SPL at one meter.

Condenser microphones: 3% at 150 dB SPL, 1% at 148 dB SPL (Crown spec).

Distortion of short-enough duration may be inaudible. We can tolerate more distortion on drum hits than on sustained tones because the peaks of the drum hits are short in duration. For a 1-kHz tone burst, the longer the pulse length, the more audible the distortion. 10% THD is just audible for 2.5 msec duration; 5% for 7.5 msec; 1% for 13 msec, and 0.5% for 25 msec [22].

HARMONIC DISTORTION
OBJECTIONABLE
Music: 2-2.5% Speech: 3-4.4%

TOLERABLE
Music: 1.35-1.8% Speech: 1.9-2.8%

PERCEPTIBLE
Music: 0.7-.75% Speech: 0.9%

JUST-DETECTABLE 2ND HARMONIC DISTORTION
<100 Hz: 25%
100-200 Hz: 3%
200-400 Hz: 1%
400 Hz up: <1%

JUST-DETECTABLE 3RD HARMONIC DISTORTION
<100 HZ: 5%
100-200 Hz: 2%
200-400 Hz: 1%
400 Hz up: <1%

JUST-DETECTABLE CROSSOVER DISTORTION
One tone: .15%
Two tones: 2.5%
Three tones: 4%
Complex music: 6%

Figure 2. Audibility of various kind of distortion.

NOISE

Noise is unwanted sound, such as tape hiss or console noise. Noise is inaudible if it lies outside the audible band, is below the threshold of audibility, or is masked by the program.

As mentioned earlier, the ear is less sensitive to low-frequency sounds than mid-frequency sounds. Consequently, low-frequency noise is less annoying than high-frequency noise. To make noise measurements correlate better with noise audibility at various frequencies, the noise signal is often measured through an A-weighting network that shapes the frequency response much like that of human hearing.

The signal-to-noise ratio (S/N) is the ratio in dB between the signal voltage and the noise voltage. A S/N of 60 dB is adequate in the presence of signal because the signal masks the noise. During pauses in the music, a S/N of 70-80 dB generally results in inaudible noise. The required S/N depends on the loudness of the playback and the quietness of the listening room [23].

Digital recordings of 90-to-98 dB S/N and analog recorders with 4-band noise reduction have inaudible noise at normal listening levels [24].

When you combine noise sources of equal level, the noise increases 3 dB for every doubling of noise sources. This fact is important in multitrack recording. Every time you double the number of tracks mixed in at equal levels, the noise goes up 3 dB. So, to reduce noise, it helps to mute tracks that have nothing playing at the moment.

For example, suppose a recorded tune starts with a bass solo. When you mix this tune, you should mute all the other tracks during the bass solo, then switch on the tracks for each of the other instruments just before they begin playing.

For a 16-track mix, the S/N of each track must be 87 dB for a final S/N of 75 dB, assuming all tracks are mixed in at an equal level. If they are mixed at different levels (normally the case), then a 32-track mix requires approximately 87 dB S/N on each track to produce a mix with a S/N of 75 dB [25].

PHASE RESPONSE

The *phase shift* of an audio component is the difference in phase angle between its input and output signals. *Phase response* is phase shift vs. frequency.

A delay between input and output that is constant with frequency produces a phase shift proportional to frequency. That is, a linear phase response (not necessarily flat, but rising with frequency) is the result of a delay. This kind of phase shift is inaudible.

According to Lipshitz and Vanderkooy, a nonlinear phase response (phase distortion) is audible, but is so subtle that it is practically insignificant in audio [26].

Clark had these findings about phase distortion: When phase shift, time dispersion, and frequency response are all changing simultaneously, what we hear is the change in frequency response. Since phase distortion is hard to hear, but frequency-response deviations are obvious, methods that improve frequency response should be used, even at the expense of phase linearity [27].

GROUP DELAY

Group delay is the rate of change of phase with frequency. If phase response is linear, group delay is constant with frequency, and is inaudible. Group delay distortion is the presence of different group delays at various frequencies.

There is some disagreement about the audibility of group delay distortion. According to Fincham, group delay distortion at low frequencies can produce subtle but clearly audible changes in sound quality [28]. According to Clark, 7 msec of group delay at low frequencies is inaudible [29]. This value greatly exceeds that found in high-fidelity equipment.

ABSOLUTE PHASE (ABSOLUTE POLARITY)

If a musical instrument produces a positive sound pressure at the onset of a transient, and a speaker reproducing this instrument also produces positive sound pressure at the onset of the transient, the speaker is in correct absolute phase. In other words, the polarity of the speaker's sound pressure matches that of the original source.

A change in absolute polarity is audible but subtle. Lipshitz and Vanderkooy report that, "Particularly on percussion signals there is a change in perceived depth, high-frequency detail, clarity, and level [29]."

To make sure your monitor speakers are in correct absolute polarity, put a microphone in a kick drum, monitor the mic, and have someone beat the drum. The woofer cone of each monitor speaker should move toward you at the instant the drum is hit.

DYNAMIC RANGE

Dynamic range is the difference in decibels between the loudest and quietest sound a source can make. According to Fielder, a dynamic range of up to 106 dB is necessary for noise-free reproduction of music

with a white-noise floor [30].

Where did he get the 106-dB figure? The maximum output level of consumer speakers is about 110 dB SPL, and the minimum audible level of white noise is 4 dB SPL. 110-4=106 dB SPL required dynamic range.

The ambient noise level of listening rooms is 40 to 50 dB, but we can hear white noise below the ambient noise floor. Fielder says that the average white-noise threshold is 4dB SPL +/- 6dB in quiet home listening rooms or studios. Low-frequency ambient noise doesn't mask the 3-7 kHz region of white noise. Also, the noise comes from the same direction as the source, while ambient noise comes from all around, so our binaural hearing discriminates against the ambient noise.

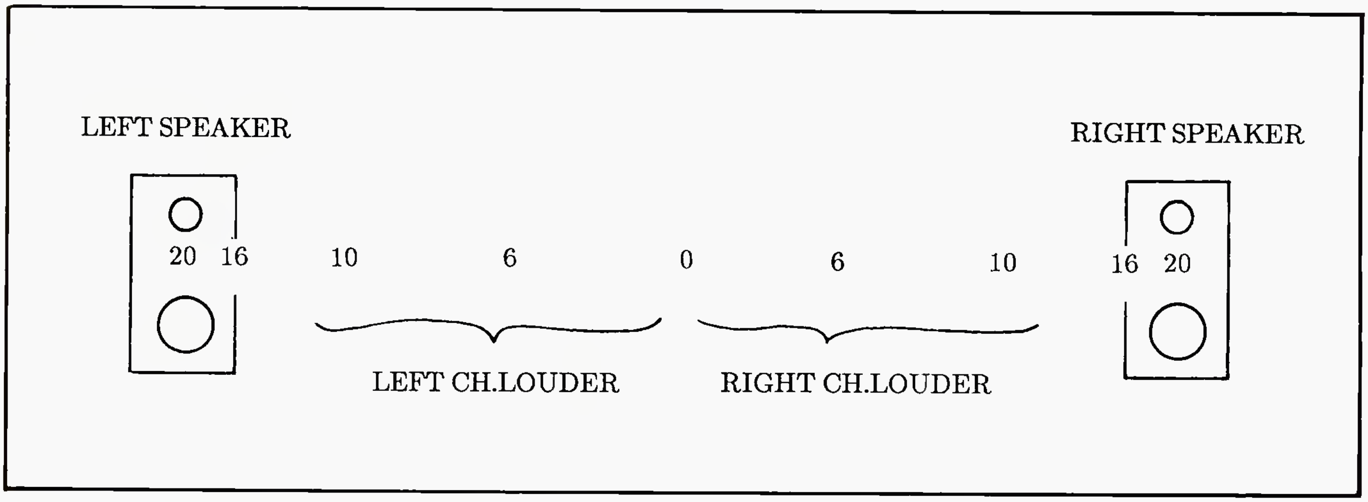


Figure 3. Approximate image location between loudspeakers vs.intensity difference between channels, in dB. (Listener's perception; listener sitting centered between speakers at a 60-degree listening angle.)

Other researchers report a maximum necessary dynamic range of 80 dB. To get this figure, they subtracted the ambient noise floor of a concert hall (35 dB SPL) from the maximum output of an orchestra (115 dB SPL in the front seats) [31]. A dynamic range of 80 dB is easily handled by digital compact discs with a dynamic range of 90 to 98 dB.

TEMPORAL INTEGRATION

This phenomenon can be described as follows: The longer a transient lasts (up to 200 msec), the louder it is. Beyond 200 msec, loudness is constant with duration [32].

According to Moir, our ears require 10 to 20 msec to judge the spectral balance of a mid-frequency tone, and 40 to 60 msec to judge spectral balance of low-frequency tones [33].

TEMPORAL FUSION

If we hear two identical sounds separated by a delay, the two sounds fuse into one if the delay is less than 30 to 50 msec [34], [35]. In other words, a direct sound and its delayed replica blend into a single sound if they are heard within 30 to 50 msec of each other. This is called *temporal fusion*. Once the spacing exceeds 50 msec, a discrete echo is heard.

Actually, fusion time ranges from 4 msec (transient clicks) to 80 msec (slow organ or string passages) [36]. You can hear this effect by setting a digital-delay unit to make a single repetition, and adjusting the delay control.

As a consequence of temporal fusion and temporal integration, sound reflections within about 50 msec of the direct sound from a loudspeaker fuse with the direct sound and increase its loudness. This loudness increase varies with frequency, because the intensity and phase of the reflections also vary with frequency. Consequently, reflections arriving within 50 msec after the direct sound affect the perceived frequency response of a loudspeaker.

In other words, sound reflections within 50 msec of the direct sound affect the perceived tonal balance. That's because the reflections make the speaker louder by different amounts at different frequencies.

So, when measuring a loudspeaker's frequency response in a room, we should measure the response effects of reflections within 50 msec of the direct sound (but no longer). This measurement will correlate better with what we hear than a simple RTA/pink-noise measurement, or an on-axis anechoic measurement. Such measurements can be made by FFT analyzers or Time Delay Spectrometry analyzers, such as the Techron TEF System 12.

The stronger the reflections, and the closer in time they are to the direct sound, the greater are their effects on perceived tonal balance. The ear is most sensitive to reflection delays around 2 to 5 msec [37]. So, to reduce the audibility of wall reflections, place monitor loudspeakers at least 3 feet from the wall behind them, or flush mount them in the wall.

The *Haas Effect* or *Precedence Effect* states that, if a sound is delayed about 20 msec and is played from a different source than the original, the ear will localize the sound at the original source. Here's a practical application: In a sound-reinforcement system with distributed speakers, some speakers are placed close to the audience. If the signal driving these speakers is delayed 20 msec after the direct sound from the stage, listeners will localize the sound at the stage rather than at the nearest loudspeaker.

STEREO LOCALIZATION

When you're listening to a stereo program over a pair of loudspeakers, you hear different instruments in different left-to-right locations between the

speakers. That is, you hear phantom images of the instruments at various positions.

If you send an identical musical signal to both speakers, and sit equidistant from them, you'll hear an image of the signal centered between the two speakers, straight ahead. If you now turn down the right channel a few dB, the image will appear closer to the right speaker. By turning down the right channel 20 dB or more, you'll make the image appear at the right speaker.

In other words, by creating an amplitude difference between channels, you've shifted the image off-center. *Figure 3* shows the dB difference between channels and the corresponding perceived image location between speakers [38].

A pan pot in the mixing console performs this function at the turn of a knob. By varying the signal level sent to both channels, it places the image of a instrument far left, far right, or anywhere in between. Stereo recordings made with coincident directional microphones work on the same principle to create a stereo effect.

Image location also can be controlled by inserting a delay in one channel. The greater the delay (up to 1 or 2 msec), the farther off-center the image appears. *Figure 4* shows the time difference between channels and the corresponding perceived image location between speakers [39]. Image locations produced solely by time differences are relatively vague and diffuse.

Stereo recordings made with spaced microphones operate mainly on the time-difference principle to localize images.

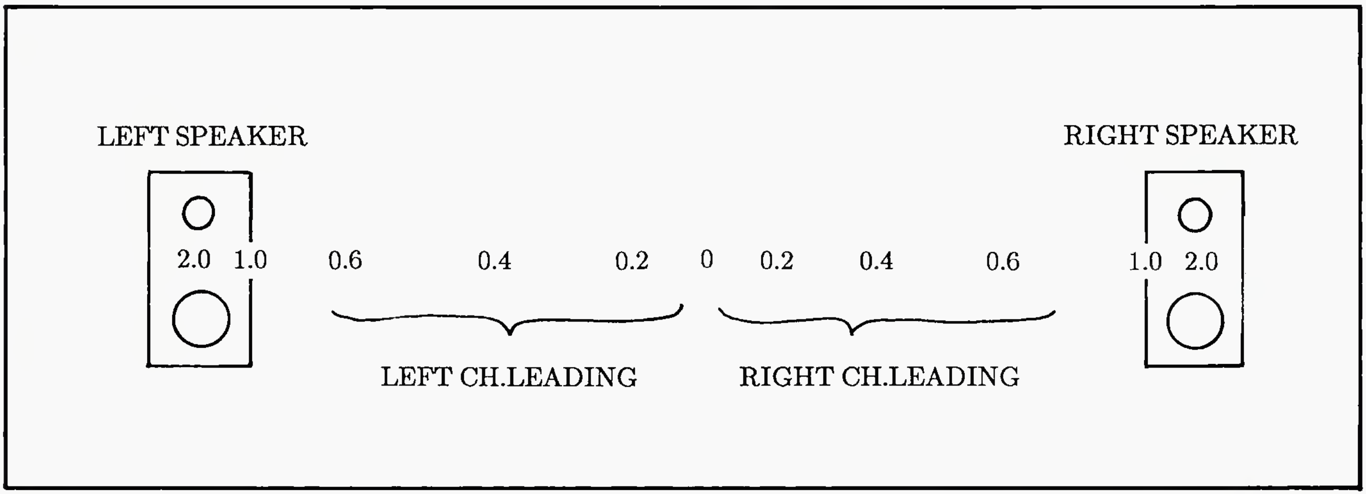


Figure 4. Approximate image location between loudspeakers vs.time difference between channels, in milliseconds. (Listener's perception; listener sitting centered between speakers at a 60-degree listening angle.)

Here are some suggestions to ensure sharp, well-defined images. Speakers mismatched in frequency response, phase response, or polar pattern can degrade image sharpness; so be sure speakers are well matched. Another cause of image vagueness is early reflections from walls behind and to the side of the speakers. Cover the walls with muslin-covered thick fiberglass insulation or Sonex. Flush-mount the speakers in the walls or space them at least 3 feet from the walls. Sitting off-center (not equidistant from the speakers) introduces a delay in one channel which diffuses the images. Sit exactly equidistant between the speakers, about as far from them as they are apart.

LISTENING TESTS

People can be trained to hear extremely small differences in audio stimuli. Trained listeners detect differences better than untrained. But some measurable effects are below the threshold of perception.

To conduct reliable listening-test comparisons of two audio devices, you must hold constant all factors which may affect the outcome, except the one being studied. Otherwise, variations in results may be attributed to the wrong factor [40]. For example, if you're trying to hear the difference in sound quality between two amplifiers, you must match their frequency responses within 0.1 dB. When you're comparing two loudspeakers, match their levels as closely as possible, because the louder speaker nearly always sounds better.

With a single-blind test, the listeners do not know how the test factor is being varied. With a double-blind test, neither the listener nor the person conducting the test know. Blind tests are necessary to remove listener bias, which easily influences perception. We tend to hear what we want to hear [41].

According to Lipshitz and Vanderkooy, electronic devices sound identical when they are driven in their linear range and they are EQ'd within 0.2 dB of each other [42].

Toole reports that accurate subjective measurements are possible, but they require multiple listeners or multiple tests with controlled variables. Informal, uncontrolled listening tests are unreliable [43].

CONCLUSION

Psychoacoustics provides highly valuable information for audio engineers. For example, if we know the frequency range and dynamic range of human hearing, then we can accommodate those ranges with similar ranges in audio systems.

If we know the just-detectable levels of distortion, noise, response errors, etc., we know what level of perfection to aim for in designing or choosing audio equipment. Total system error below audibility is the goal. Since the distortions in all the system

components can be cumulative, it's good to keep all distortion as low as possible.

If we understand masking and equal-loudness contours, we can make better mixes.

If we know the kind of loudspeaker laboratory measurements that correlate with listener fidelity ratings, then we know what specifications to look for in choosing an accurate monitor system.

If we know how room reflections affect perceived tonal balance and stereo imaging, we know how to make in-room measurements that correlate better with perceptions.

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db Test

Len Feldman

Applied Research & Technology DR1 Digital Reverberation System

GENERAL INFORMATION

THE AUDIO AND RECORDING industry has certainly come a long way since the days when *reverb* meant interposing a mechanical spring with a transducer at each end in the audio signal path. Applied Research & Technology, Inc. (ART) amply demonstrates that fact with their DR1. This versatile unit is a high-definition signal processing device that provides natural reverberation as well as creative reverb and delay effects that would not normally be found in any *real* environment. A microcomputer in the DR1 controls the high speed signal processor and allows quick and easy adjustment of relevant reverb parameters such as room size, *liveness*, color and depth. Microcomputer control also permits redefinition of front panel controls when creating special effects and allows for future expansion of effects. In fact, the owner's manual supplied with our sample, though written for the *1.0* version of the DR1 software, has an extensive addendum at the back of the book because, in fact, the software in our sample has already been upgraded to version *1.2* which offers even more flexibility and diversity of settings than did the earlier version.

The front panel controls, many of which are duplicated on a supplied wired remote control, allow you to access and modify all relevant reverb parameters easily. Once you achieve a desired sound or effect, you can store the settings for later recall in one of a hundred user presets! Alternatively, you can choose one of the factory pre-programmed presets. All presets are non-volatile; they don't disappear when you turn off the power and power up again later.

The DR1 is easily incorporated in line with consoles or musical instruments. Left and right reverb controls, in conjunction with the mono/stereo inputs and outputs are useful when no other mixing or level adjustment facilities are readily available. A rear panel *dry kill* switch removes the dry signal from both outputs (leaving only the reverb component) when external mixing is desired.

An important feature of the DR1 is called MIDI. It stands for Musical Instrument Digital Interface. This feature provides a means whereby musical instruments, drum machines, sequencers, computers, etc. may be connected to send messages to one another. We didn't get into the testing or evaluation of this feature because we did not have the necessary secondary equipment with which to interface this feature, but from its description in the owner's manual we suspect that anyone familiar with this type of interface will find the MIDI system incorporated in the DR1 to be extremely sophisticated and versatile.

The ART DR1 digital delay.

CONTROL LAYOUT

Perhaps the best way to appreciate the versatility of the DR1 is to understand what each of its front panel controls accomplishes, so we'll go over them, one by one. A red colored push button, labeled *STORE*, near the left end of the standard rack-mountable front panel, when pressed, stores all the parameters in the *Value* section of the panel into a chosenpreset, providing a nearby indicator light, labeled *LOCK* is not lit. When the *LOCK* light is on, that means that the chosen preset cannot have its values modified and then stored without unlocking it first. The push button used to unlock a preset is purposely recessed behind the front panel to prevent accidental locking or unlocking of your presets. Forty factory presets, labeled F0 through J9 are permanently stored in the DR1's ROM (Read Only Memory) along with the rest of the DR1's instruction set and cannot be modified by the user. The Table in *Figure 1* lists only the first thirty of these factory presets, along with the intended application or room description for each, since ten more (J0 through J9) were added for the revised software version contained in our sample.

Presets 00 through 99 are user presets and can be modified, stored, recalled, locked and unlocked. Presets can be transferred by recalling a preset using the RECALL button located next to the *STORE* button. UP and DOWN push buttons quickly move the display from one preset number to the next, in either direction.

The *VALUE* section of the front panel consists of five push buttons each surmounted by a two-color LED. Thus, ten parameters can be adjusted for each

preset. The five parameters that are adjustable when the LED light is green are *ROOM, PRE-DELAY, DECAY, HIGH-FREQUENCY DAMPING* and *POSITION*. When the green light under *ROOM* is lit, the display will indicate which room type is active. The DR1 implements a different algorithm for each of its room types, as described in *Figure 1*. In conventional recording practice, a delay is often inserted between the console and the reverb chamber to add an apparent depth to the reverb sound and to separate, in time, the initial sound from the dense reverberation. When this parameter is selected, the

APPLICATIONS

PRESET	ROOM	PRE DELAY	DECAY	H.F. DAMPING	POSITION	K/I MODE	DIFFUSION	MIN DECAY	DESCRIPTION
F0	EF.1	17	2.0	1	5	1	7	2.0	CHARACTER
F1	P1.0	0	0.5	0	2	1	0	0.5	SUPER TIGHT PLATE
F2	P2.0	3	0.7	0	3	1	2	0.7	TIGHT PLATE
F3	P3.0	20	1.0	1	5	1	4	1.0	MEDIUM PLATE
F4	P3.0	10	1.2	1	5	1	6	1.2	OPEN PLATE
F5	P5.0	15	1.6	2	7	1	5	1.6	LARGE PLATE
F6	R1.0	10	1.0	5	4	1	7	1.0	LIVING ROOM
F7	R2.0	15	1.8	2	3	1	3	1.8	MEDIUM BRIGHT ROOM
F8	R3.0	21	2.3	10	6	1	7	2.3	LARGE SMOOTH ROOM
F9	R4.0	5	2.8	6	3	1	4	2.8	LARGE EMPTY ROOM
G0	R5.0	37	3.2	8	7	1	7	3.2	LARGE NIGHT CLUB
G1	H1.0	29	2.5	2	5	2	4	2.5	PRACTICE HALL
G2	H2.0	40	2.0	6	7	2	7	2.0	NATURAL HALL
G3	H3.0	50	4.0	9	3	2	3	4.0	MEDIUM HALL
G4	H4.0	57	3.3	5	6	2	7	3.3	LARGE HALL
G5	H5.0	49	5.4	6	6	1	7	5.4	EVENT HALL
G6	EF1	21	5.2	3	6	1	7	5.2	CAVERN
G7	EF1	0	12.5	0	5	1	2	12.5	CANYON
G8	EF2	22	4.2	1	15	1	1	0.5	FLANGE
G9	EF2	11	2.4	1	15	1	1	1.5	STEP FLANGE
H0	EF2	64	2.0	2	10	1	1	1.0	DRONE
H1	ER0	99	0.2	2	-	1	6	-	REVERSE SLAP
H2	ER0	111	1.2	0	-	1	5	-	REVERSE SWELL
H3	EG0	43	1.1	0	-	1	4	-	GATED ECHO
H4	dd1	1.00	1.00	0	0	3	-	0.50	PING-PONG
H5	dd1	0.11	0.10	10	16	1	-	0.09	ECHOREC
H6	R5.0	20	3.7	4	4	1	7	3.7	POP ROOM
H7	P3.0	7	6.2	13	8	3	5	6.2	TUNNEL
H8	R4.0	12	25.0	3	4	3	7	1.3	DYNAMIC ROOM
H9	R4.0	0	5.0	0	5	1	7	0.3	DYNAMIC PUMP

Figure 1. This table shows factory presets of the DR1. (Ten more presets have been added in a software update that has been released since this chart was published.)

amount of pre-delay can be adjusted and is displayed. The *DECAY* function, of course, determines the time required for the reverberant sound to decay to a -60 dB level and as this parameter is adjusted, decay time is shown in seconds in the display area. High frequency damping helps to determine the softness or liveness of a room and when this parameter is chosen to be adjusted, relative values from 0 (no H.F. damping) to 19 are available and are displayed as they are selected by the *UP* or *DOWN* buttons beneath the display. The last of the five buttons determines the listener's *POSITION* when the associated LED is green.

Changing the value here, over a range from 0 to 9 as shown in the display, varies the mix of initial sound and later reverberation placing the listener anywhere from at the front of the room to the back of the room.

These same five buttons, when depressed a second time, cause the LED's to change their color to red, at which time, alternate functions are parameters for adjustment. The first of these is called KILL/INF or K/I. Its most usual setting is *1* . When set to this mode, depressing a separate KILL/INF push button elsewhere on the panel kills the reverberant signal entirely. In Mode 2, the same K/I push button kills the decay position of the signal but early reflections are still enabled. Finally, if number *3* is brought up in the display, the unit is put in the infinite hold mode. This effect is not unlike that of the *sustain* pedal on a keyboard. The next button, when pushed a second time, allows adjustment of *DIFFUSION*, varying the reverb sound from rough to smooth by increasing echo density and filling in the spaces between individual echoes. The third buttons' alternate mode is *MIN DECAY*. By setting the minimum decay for a short decay time and the regular decay for a longer time, sound in a normal program will quickly fade away at the higher signal levels, maintaining a clean sound. During quiet pauses, or at the end of the material, the decay time will revert to the maximum value originally set by the DECAY adjustment. This type of dynamic, or changing reverberation does not occur in nature, but can be used to create a distinct reverberation free of the masking din that can accumulate with longer decay times.

The last two buttons, in their alternate (red LED illuminated) modes are labeled MIDI PGM and MIDI CHAN and have to do with the MIDI functions we mentioned earlier, but did not test. To the right of this elaborate *VALUE* adjustment section is the LEVEL adjustment section of the DR1 where we find an LED bar graph that shows input level, calibrated in 3 dB steps from -21 dB to 0 dB and left and right reverberation slider level controls. At the top of the LED bar graph is an OVF (overflow) indicator which warns when numerical values in the digital signal processor exceed the processor's calculating range. In any digital audio signal proces-sing system, overflow will cause audible distortion and flashing of the OVF LED means that input levels must be reduced.

The rear panel of the DR1 is equipped with left and right channel input and output 1/4-inch phone jacks, a two position input label switch, the previously mentioned DRY KILL switch, a telephone module type jack to which the remote control cable is attached, input and output multiple-contact DIN sockets for the MIDI feature and a KILL/INF jack that duplicated the function of the KILL/INF push button on the front panel. The HIGH setting of the input level switch selects an operating range of from 0 dB to +12 dBV maximum input, while the LOW setting chooses a range of from -12 dB to 0 dBV maximum. A bass roll-off switch is accessible if the top cover of the DR1 is removed. This switch selects between two roll-off frequencies: 50 Hz and 150 Hz. Units are shipped with this switch set to the 50 Hz roll-off position.

A TECHNICAL DESCRIPTION OF DIGITAL SIGNAL PROCESSING IN THE DR1

Although the DR1 is predominantly digital, it must, after all, interface with analog audio signals. The input amplifiers bridge a balanced line and provide buffering between the audio source and the DR1's internal circuitry. Next, input filtering removes unwanted high-frequency material. The signal is then sampled at discrete instants of time and converted into a continuous stream of digital numbers by the analog-to-digital converter. After conversion, the *numbers* are stored in memory.

A high-speed digital signal processor capable of performing millions of arithmetic calculations per second, is at the heart of the DR1. It retrieves the encoded numbers that represent the input signal from memory and processes them according to the selected parameters. The calculated results are then

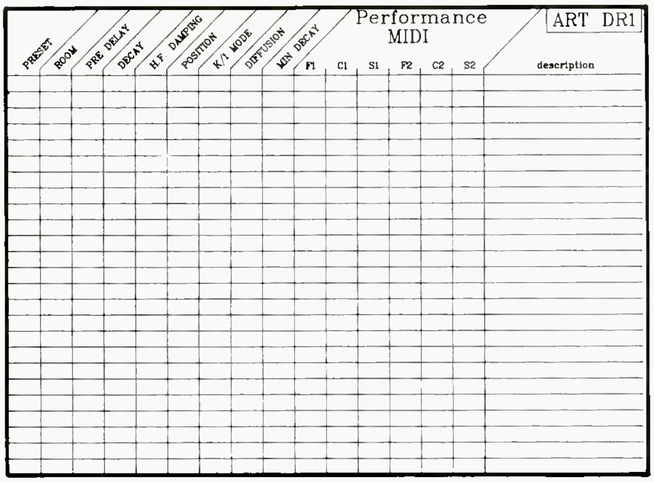

Figure 2. Sample worksheets such as this are supplied with the DR1 to help the user save information about created presets.

stored back into memory. They represent the original signal with reverberation added.

At regular specified intervals, this processed data is recalled from memory and converted back into an audio signal by a D/A (digital-to-analog) converter. Alternate samples are fed to the left and right output channels. Output filters remove any high frequency noise introduced by the sampling process. Finally, the output amplifiers buffer the signals and provide line driving signal levels at the outputs of the unit.

The microprocessor, along with its operating software (in EPROM), monitors the front panel controls, MIDI interface, REMOTE control and the rear panel external control jack for the user input and output setting information. Button depressions are translated into commands understood by the digital processor. The microprocessor also controls storage of front panel settings in the preset memory and their retrieval. A lithium battery preserves the presets as well as current front panel settings when AC power is turned off. According to ART, the battery can be expected to last for about ten years.

LAB MEASUREMENTS

A complete table of VITAL STATISTICS covering the manufacturer's stated performance specifications and our own confirming measurements and observations will be found at the conclusion of this report. For the most part, our measurements were minimal, since there is not much that really can be measured *on the test bench* when it come to a unit of this type. Suffice it to say that all of the usual audio measurements such as bandwidth, signal-to-noise ratio or dynamic range were at least as good as claimed if not better. There was absolutely no change in performance after several hours of use of the equipment, although the unit itself did run warm to the touch. ART mentions this fact in their owner's manual and suggests that the DR1 be given adequate ventilation when installed; at least one inch above and below the unit for adequate convection cooling.

COMMENTS

The DR1 is, by far, the most sophisticated and effective digital reverberation signal processor that I have encountered. Reverberation effects can be altered with extreme subtlety or over a very wide range that should satisfy the requirements of the most demanding professional audio practitioner, whether that practice is limited to recording studio work or extends to sound reinforcement and other audio applications. The number of variations that you can achieve with the DR1 is almost limitless and anyone purchasing this unit will want to experiment with it extensively in order to get a feel for what the unit can do. With so many possible setups and presets available, ART has wisely included a couple of handy worksheets, onto which you can list the various parameters of the presets that you yourself create with the unit. A size-reduced copy of such a worksheet is shown in *Figure 2* to give you some idea of how it is organized. Of course, you can easily copy this worksheet once you own the DR1, if there aren't enough spaces on it for all the presets you want to create. Remember, you can create 100 of your own presets over and above the 40 factory presets that are built into the unit.

Developments such as the DR1 point up the fact that digital signal processing is useful not only as a straightforward means of audio recording but that it can serve other needs which were formerly met by analog and even mechanical devices (spring and plate reverb and time delay units)that seem crude when compared with products such as the DR1. Not only do such digital signal processors offer much more versatility and flexibility than did their analog predecessors, but what's far more important, the quality of the audio signals created by such digital devices are cleaner than anything achieved with mechanical *springs* or more recent analog time delay units based upon *bucket brigade* devices. Furthermore, units such as the DR1 nearly approximate (though still not completely so) the sorts of sound reverberation that we experience in real life while allowing the creative audio engineer to produce effects that were simply not possible in the days of mechanical reverb units.

VITAL STATISTICS
DIGITAL REVERBERATION SYSTEM

MAKE & MODEL: **Applied Research & Technology Inc. DR1**

SPECIFICATION	MFR'S CLAIM	db MEASURED
Maximum Input Level	+12 dBV	+12 dBV
Maximum Output Level	+12 dBV	+12 dBV
Operating Levels		
HIGH Setting	0 dBV	0 dBV
LOW Setting	-12 dBV	-12 dBV
Input Impedance	47k ohms,bal.	Confirmed
Output Impedance	1 k ohms	Confirmed
Bandwidth		
Unprocessed(Dry)	35 kHz	37 kHz
Processed(Reverb)	14 kHz	13.5 kHz
Dynamic Range	<90 dB	94 dB
D/A Converter	16 bits,linear	
Power Requirements	105-125 V AC, 50-60 Hz	
	25 watts	22 watts
Dimensions (HxWxD,in inches)	1-3/4x19x9	Confirmed
Suggested Price	$1,295.00	

The Following Specs are Software Dependent

Decay Time	0.1 to 25.0 seconds*	Confirmed
Pre Delay	0 to 200 msec.*	Confirmed
Factory Presets	40 (F0 to J9)	Confirmed
Room Types	21	Confirmed
MIDI Receive Channel	1-16,OMNI on/off	Not Tested
MIDI Program	0-127	Not Tested

*Different in some *effects* programs

 ■

Brooke Sheffield Comer

Mastering Digital Broadway

Here's the story of a recently released album that was both musically and technically produced for the digital disc market.

VIRTUALLY AS SOON AS CD's hit the market astute mastering engineers realized that the new medium had unique advantages over vinyl. But they would go untapped without a special CD, which could utilize the increased dynamic range to a maximum potential. Producer Mike Berniker (whose work on Barbra Streisands's *People* won an Grammy) and Grammy nominee Byron Olson too the notion a step further. Why not, Berniker asked himself, release a unique CD project, especially arranged, composed, and recorded to bring out the most sophisticated dynamics of compact disc technology? Their end product, *Digital Broadway*, engineered by Keith Grant at London's CTS studios, is a testimony to their success.

CREATING THE ALBUM

Berniker began to plot *Digital Broadway* a year ago. "I'd been working on remixing and combing various CD projects from catalogs," he explains. "Of course, when you remix from analog multitrack to digital two track, you realize that there are natural dynamics in the course of the original recording that are more evident in the new form. The nature of the medium has a greater potential to reveal sound." Plus, he realized, he could mix digitally for CD with much more flair. Pinning and leveling problems no longer created restrictions. He found he could find subtlety that were not in the original mix. While experimenting with different tracks in the catalog Berniker realized that no one had yet come up with a musical conception that utilized this dynamic conception.

When Berniker proposed his idea to his peers, the response was guarded but enthusiastic "Not long ago, no one in the business knew that the CD would become such an important factor in the consumer market, which is the reason why my idea hadn't been done before," he notes. "I knew that I could make music sing in a way that hadn't been done before, on vinyl, but I had more to do than just come up with the idea. I had to conceptualize the creation of the project, step by step." Berniker knew that if the project wasn't; done just right, it wouldn't make it on the market. It had to be dramatically different.

Byron Olson, the composer/arranger/orchestrator for an array of stars including Perry Como, Peggy Lee, and Johnny Mathis, was the logical person for Berniker to turn to. The two worked hard for ten years, and enjoyed a musical sympatico that is rarely found between producer and composer. "I knew Byron could do the job, because he understands both orchestrative needs and my lingo," says Berniker. And a mutual understanding was essential. Each song selected for the repertoire(which includes 14 Broadway masterpieces), was chosen for specific tempos, a variety of crescendi and diminuendo.

SPECIAL MUSIC FOR THE CD

Because the unique dynamics of CD recordings can reach as much as a 40 dB variance, Olson felt that the whole idea of a recording based on use of the full spectrum of dynamics was in demand. "The notion seemed totally musical and entirely possible,

Mike Berniker at CTS Studios in London.

now that digital recordings have come of age," he explains. In classical music, this type of recording has always been done to demonstrate the clarity, quietness, and strength of CD sound. The objective in *Digital Broadway* was to do the same thing with popular music."

Olson was optimistic; enthused that Berniker's idea would be exciting to execute in terms of a musical idea, he set out to design an arrangement to provide an even greater dynamic range. "There are spots in the recording where there is no music at all," he points out,"just chunks of silence to demonstrate the quietness of digital CD recording."

Subtle sonic differences, more easily discerned when recorded especially for CD, are just one of the marketing factors that Olson and Berniker took into account. "It's quite unusual to find such a dramatic change in dynamics," says Olson. "The sound

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in pop music is usually either very soft or very loud. It's rare that you come down to a double piano, then in a few bars you're at a triple forte, which is the other end of the spectrum. This kind of input never has the same impact on vinyl."

NEW HORIZONS FROM THE CD

Working with such drastic extremes of the sound spectrum opened new horizons for Berniker and Olson, even though the two come from diverse musical backgrounds. The beauty of *Digital Broadway* was the unique potential for innovation that it offered us," says Berniker. "And as a result, just working on the project opened new doors for both of us. I know that no one could possibly make an analog master with the dynamics and dimension that Byron has written out in these arrangements. The whole idea had to come out of musical production experience, which gave us an edge. Only someone who has lived with the process of writing music all his life doesn't have to worry about a plus eight. I can now do a plus 50 dB and not be concerned about pinning and distorting, or not being able to cut the master. Before *Digital Broadway*, I'd been inhibited, but the project opened up a whole new vein in terms of what I can now accomplish."

Olson too was able to express himself more fully, using the broad range unique to CD's "not only in the total freedom of expanded dynamic dimension, but in the actual notes themselves," he reveals. "Mike is not the kind of producer who tells you what to write, so I had that extra outlet. I find it terrifically exciting to compose using broad dynamics. It's something that's always been done in classical music but not in any other genre." Olson and Berniker are also working on a *Digital Hollywood* project, and have recently recorded a *Digital Christmas* compilation, scheduled for December release.

WORKING WITH A LARGE ORCHESTRA

Conducting such a large orchestra as Orchestra Manhattan (an orchestra hand-picked for virtuosity and experience, and which includes Olson on piano), could have been difficult. "But we made sure that with an orchestra this size, it's important that everyone have a clear idea of what they're playing," Berniker observes. We made sure they observed the markings, played at a general volume level, and as a result we didn't have to worry about mixing later. We went for the dynamics of the room. We recorded what was was there, the live sound, knowing that the spontaneity would come through."

When Berniker sat down at the mixing board, he found that the process was less complex, with fewer overdubs. "Keith told me I was doing the opposite of what I'd instructed him to do," Berniker laughs. "Since the project was recorded digitally, and all of Byron's markings were played to his exact specifications, the mix simply involved spreading out the sound and doing some imaging. The most exciting part was going for the ultimate sound during the performance, rather than coming back and overdubbing a flute track."

LOOKING TO THE FUTURE

Berniker, with his heightened expertise in digital, finds his later work more dramatically flavored. "I tend to go for things I wouldn't have gone for before," he notes. "I was always pushing everyone's range, and getting the contra-bassoon, which is the lowest instrument in the orchestra, to play a double b flat, the lowest in the orchestra, to E, two octaves above middle C on the piccolo.

I think those broad range attempts really show up the find quality of a CD recording, when there is no distortion from any level in terms of transition. That's the wonderful part of Byron's brilliance. It's one thing to talk about markings and another to talk about range difference in the course of four bars."

Yet the three and a half weeks of around the clock work that Olson and Berniker put in on *Digital Broadway* was an attempt at more than simply reaching a new technical plateau. "We're not just trying to create things that are appealing from a sound effect standpoint, so that we can show off a mastery of dynamic range," Berniker assures us. "We're taking the music in the sound to a new extent." The fact that Olson and Berniker are musicians themselves gives them a special advantage. "Record company executives can't just put in a call and order us. A talent like Olson is rare indeed. One of the things I find so satisfying about working with him is the jazz chops he brings to harmonize familiar material. That way, he can constantly refresh what could be very dull stuff." Rhythmic re-thinking, after all, is the domain of only the most talented arrangers.

THE ALL-DIGITAL STUDIO

CTS, the first all-digital recording facility in the world, seems the perfect choice for Berniker and Olson to have made when they sought a studio for *Digital Broadway*. The CTS board, with its 48 inputs and 32 outputs, and internal dynamic range of 192 dB, works in conjunction with the Sony 3324 digital multitrack and the Studer A 800 24 track machine. "But we found that the music defines what we do in the studio rather than the gear in that studio," says Berniker. "The digital board was wonderful, enabling us to go from digital source right to digital machine. But in terms of dynamics and illustrating what we needed to accomplish, not even the most finicky listener can detect the difference between an all-digital and an analog board." Still, Berniker gives the digital board credit for a cleaner sound in the transcript. "But I like the warmth of analog, too," he adds.

Berniker gleaned a significant portion of the expertise he brought to *Digital Broadway* from his production work with the artists including Barbra Streisand. "If I did nothing else but let Barbra be as daring as she could be, I was a good producer," he reveals. "And the same principle exists here. To do the a project like this properly, you have to be daring and go all the way. Most people in the recording business, when working with a new concept for the first time, can't do that." As a result, Berniker believes CDs are renowned for their clean sound, while other potentials remain untapped. "Something clean and dull doesn't make it for me, he says, unless you have something that's clean and more expressive as a result of your own ability to use the medium." Given Olson and Berniker's pursuit to fill the void that technology has left in its wake, the future for expressive CDs is great. But rather than exalt the technology, they'd rather utilize the technique to invigorate music. ■

Gene Juall

The Ramsa WR-8428 Console

Here's a brand-new multi-purpose console that you may well find exactly serves your needs.

Ramsa WR-8428 Console

THE RAMSA WR-8428 WAS DESIGNED to meet and anticipate the varied and rapidly evolving requirements of today's post-production and broadcast industries. This new 24-track console is essentially an expanded version of our current post-production mixing console, the WR-8616. Because the 8616 was Ramsa's first product for the professional post-production/broadcast markets, we started out on a relatively modest scale. But with the success and overall acceptance of the 8616 in the market, the need for a more powerful product soon became clear. Accordingly, the 8428 offers 28 input channels and 24-track monitoring capability, as opposed to the 16-in/16-track capability of the 8616. As we'll see, it offers many other design refinements as well.

One key design goal was to make a console that would allow a post-production video house to move up from an 8616 to an 8428 and not have to spend a lot of time re-educating themselves. Our plan was to make the 8428 as operationally similar to the 8616 as possible.

In achieving that goal, we received quite a bit of input from end users. It started out on trade show floors, where we demonstrated the 8616. People expressed an interest in a bigger version of the console and made suggestions. We sat down with all interested parties and got their ideas on what the 8428 should do. We visited a lot of our key dealers and production houses. Many of them already had 8616's, which made it easy to get their input. We also worked closely with what we considered the most advanced post-production houses. Through their input, we hope to ensure that this product will have a long lifetime in the marketplace.

The professionals we consulted identified two general areas of need. One was the necessity of accommodating Dolby 4-channel mixing, which should become more and more prevalent with the advent of innovations such as home Dolby decoding units for video playback equipment. The second request was for more flexible signal routing than ever before. In this area, an important goal for us was to provide this flexibility without making the console unit complex. Because even though audio is growing in importance, the visual element of the program remains the first priority for many post-production professionals. There is usually neither the time nor the budget to come to grips with an audio that's particularly difficult in layout and operation.

As far as performance goes, the 8428 electronics are

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generally quieter and offer more headroom than the 8616. The 8616 was the first Ramsa console to provide all balanced inputs and outputs and the 8428 also provides this feature.

A GENERAL OVERVIEW OF THE WR-8428

The 8616 distinguished itself as the first fully-modular Ramsa console. Following the same modular design philosophy, the 8428 offers numerous options for tailoring the console to different specific applications. Looking at it from left to right, the 8428 is laid out as follows: There is provision for 28 input modules, 4 subgroup modules, 2 send modules (left & right), 4 matrix output modules, 1 monitor module and 1 talkback module. Of these principal console sections, the input, subgroup and send master modules all offer options with regard to specific module configuration. At this time, the master, matrix monitor, monitor and talkback sections are all standard modules.

Most of these standard modules cover very basic mixing functions. The master module allows all of the 8428's signal buses to be routed to the console's stereo master outputs. The monitor module allows signals to be routed, in different combinations, to the control room and/or the studio. The talkback module provides a 1 kHz/80 Hz slate oscillator, a microphone input and the necessary controls for routing these signals sources to various destinations. The matrix modules have a greater variety of functions. And, by mid-1987, Ramsa will be offering an alternate version of the matrix module specifically designed for Dolby 4-track/surround mixing. These matrix module functions will be discussed in detail below. But first, let's look at the options for the input, subgroup and send master modules.

MASTER VS. STEREO INPUT MODULES

Each input slot can be fitted with either a mono or stereo input module. The customer can elect to install whichever combination of these happens to suit his particular application (including any number of "blank" modules).

The mono module has a microphone input (with a 48V phantom power supply) and a line input. These are selected via a switch near the top of the module, which also has a *tape* position. As a general rule, post-production applications are typically looking for more line level inputs than mic inputs, to accommodate sound sources coming from the tape machines, cart machines, etc. So the mono module input is also equipped with a switch that changes the microphone input to another line input.

The stereo input module provides two simultaneous line inputs, but no microphone input. Of the two options, the stereo module offers less signal processing power, because it is mainly intended for work with pre-eq'd, stereo line levels signals coming in from a VTR, audio tape machine or other audio sources.

The mono module, then, carries many of the features you would find on a music recording console. It has, for example, a +15 dB, 3-band eq section with shelving-type high and low-frequency controls, all sweepible. The eq section, which is deafeatable, also includes an 80 Hz cut button, for eliminating low-frequency rumbles and other unwanted signal material in that frequency range.

The Ramsa Model WR-8616.

The mono module also includes four auxiliary sends (Effect 1-4 and Send 1-4), which are selectable for pre- or post-EQ operation. The pre and post switches on these controls illustrate that combination of flexibility and simplicity of layout that we've found video people really appreciate. The pre and post switches enable the four auxiliary sends to perform a variety of functions (as effects sends, cue sends, or even subgroups), and they provide a neat, clean alternative to placing a separate control on the module for every one these functions.

The program address controls on both the mono and stereo modules take this concept a step further. They provide direct punch-in on subgroups 1 through 4, and a fifth button provides punch-in on the 1-4 master bus, routing the signal through the channel module's panpot. Now, the four subgroups can also be made pannable, if you wish, via a small switch on the PC board. People who do a lot of music work, as opposed to straight post-production, sometimes prefer to have the subgroups pannable at each input. What they can do is enable the group pan when they set the board up. Once it's set to preference, it can stay that way. And there's one less control on the module to worry about.

The same principle applies to the Solo button for each input module. It can be set pre- or post-fader via a jumper inside the module. Again, experience has shown us that music people generally like to have the solo post-fader, and post-production people tend to prefer it pre-fader. Neither one is apt to do a lot of switching between these two options, so the control is placed beneath the module.

One difference between the mono input module and the stereo input module is that the stereo module has two 30 dB trim controls–one for each line input–as opposed to the single 30 dB trim pot on the mono module. This makes it easy to handle level discrepancies between the left and right channels of an incoming stereo signal. It's also useful for broadcasters, who might want to use one channel of the stereo module as a redundant feed for a microwave or telephone link–so that, in case one signal goes out, the other one is there. The two trim controls allow the signals to be matched, if they happen to be at different levels. The stereo input module also includes a mono switch that sums the left and right input signals after the trim circuits. Using the two trim controls, the two signals can be balanced as required before they are summed.

As was mentioned earlier, the stereo input module does have somewhat less processing power than the mono input module. For example, it does not have the pre/post option on the Effect left-right and Send 1-4 auxiliary sends. We spent a lot of time with broadcasters and post-production people on this point. Basically, what they all told us was to make the Send pre and make the Effect post on the stereo module. Why? Because the signal is generally coming in from the cart machine or VTR pre-eq'd and pre-sweetened, so they're not going to do that much with it.

For the same reason, we designed a simplified eq for the stereo module–just shelving type high and low controls. There's also a defeat for the eq inside the module. Some people we consulted went so far as to say they never want anyone messing around with the eq on these pre-sweetened, pre-eq'd incoming signals. So they can eliminate the eq section entirely. On the other hand, some admitted that not all incoming signals are acceptable. So we provided them with an 8 kHz high cut switch and an 80 Hz low cut(as opposed to just the low cut on the mono module).

The slots allocated for the four subgroup buses and the two (left-right) send masters can be filled with the customer's choice of either a basic group module or a tape monitor module. Although the send masters are not, strictly speaking, group buses, each of them can function as a subgroup through the choice of the appropriate module. This, of course, is in addition to their nominal function as cue and effect sends.

The basic group module is the simpler of the options that can be installed in these slots. It consists of a fader, a return and a 1-4 send, which allows you to send that group output to the master section.

The tape monitor nodule, as its name implies, adds tape monitoring capabilities to these basic functions. Each module provides four tracks of tape monitoring, with two separate level controls for each track of monitoring. So if the 8428 is to be used for a 24-track application, for example, the customer would install tape monitor modules in all six of the available slots. This would provide the necessary 24 tracks of monitoring (6 modules x 4 tracks per module= 24 tracks). For a 16-track application, the customer would probably want to install four tape monitor modules, in the four subgroup slots, and two basic modules in the send master slots. An 8-track facility would require two tape monitor modules, and a 4-track facility only one tape monitor module. Facilities that deal exclusively with stereo sources (VTRs, cart machines, etc.) could install six basic modules.

One of the 8428's major improvements over the 8616 comes in the form of two multipin RS232 connectors that serve as a secondary set of input/output connectors for a 24-track machine. Combined with the tape monitor modules discussed above, these modules form a second 24-track machine.

A typical dual 24-track application would work like this: The output of the multitrack that comes in on the multipin connector goes directly to the tape monitor section on the group modules. In a post-production context, it would be a good idea to use the multipin connector for an in-house multitrack with all the library effect tracks on it. Typically, these have been eq'd and sweetened already, so there's generally no need to use the console's input modules to do a lot of processing on these tracks. Where the processing power is needed is on the raw dialog tracks and other material coming from the production house. The multitrack with that material on it can be brought in via the line inputs on the individual input channels, which will provide all the eq and processing you need.

One attractive feature of the tape monitor modules is that they provide two discrete monitor mixes for each track. One comes up on the left-right masters (which could also be called *control room masters* under certain mixing circumstances). The second mix comes up on the previously-mentioned send circuits associated with the left-right send masters (and which we could call *cue sends* under certain mixing conditions). If you're overdubbing, for example, this enables you to send a monitor mix out into the studio and set up an independent mix in the control

room. Once you get down to the mixdown state, you can use that monitor section as a return for the multitrack that has effects on it. In all, the 8428 offers powerful signal routing for a board that will sell for less than twenty thousand.

DOLBY 4-TRACK MIXING AND OTHER MATRIX MODULE APPLICATIONS

Further mixing flexibility is provided by the 8428's four matrix modules. There are inputs that can be brought up on each matrix module: The four subgroups, the send master left and right, and the master left and right. The matrix modules can be used to generate the four channels of information (front left, front right, front center and surround) required for Dolby 4-track mixing. This can be accomplished without having to use the four subgroups, which are much needed for other purposes when you're mixing as many as 28 inputs.

The matrix module slots have been wired to accommodate specialized Dolby matrix modules–tentatively scheduled for mid-1987 release–that will further facilitate Dolby mixing. The Dolby matrix module should simplify the procedure dramatically by allowing the constant monitoring and switching functions associated with a Dolby mix to be performed right on the matrix module, rather than on a remote black box.

Apart from Dolby surround mixing, the matrix modules can perform a variety of roles in the post-production environment. Suppose you have a client who envisions a number of different markets for his project and wants a digital two-track master, an analog two-track master and also a mono master. With the matrix output capability, you can now make all those masters on one pass, eliminating the need to reset the board for each pass.

This type of capability makes the matrix outputs useful in a production context as well. A large scale production– coverage of a sports event, for example–could require a stereo send to a recorder, another stereo send going out on the air and a third send going out on the air in mono. Each mix would have to be slightly different (less bandwidth on the AM radio feed, for instance), and the matrix module could be used to create the appropriate mix for each feed.

Although the 8428 was mainly conceived as a post-production recording console, a number of people have expressed interest in it as a production board, largely due to the matrix modules. To cite another typical television requirement, the matrix outputs could readily be used as mix minus outputs. In short, its price and flexibility make it ideal for use in a comprehensive, all-under-one-roof production/post-production facility, or even in industrial audio/video applications.

METER BRIDGE LAYOUT

The metering format on the 8428 follows a pattern that was very well received on the 8616. Basically, while everyone seems to prefer LED meters on input channels, we found that many people in the broadcast and production areas are more comfortable with VU meters on the outputs. Accordingly, the 8428 combines LED input metering with VU output meters. Here's how they're configured.

There are 24 input LEDs. The last four can be switched for stereo monitoring, if they are to be used in conjunction with stereo input modules, or they can be switched to provide metering for mono input channels 25-28. A select switch on each module determines whether its attendant meter is monitoring tape or the "live" input signal.

Output metering is handled by a group of eight medium-sized meters and two large ones. The large meters monitor the left and right stereo masters. Also, whenever a solo button is engaged, the right master meter automatically switches to read the solo signal. The left master meter can also be switched to read the console's mono output, which sums the left and right master outputs.

The eight medium-sized VU meters are laid out in two rows of 4 meters each. The top row reads the four subgroup outputs or the effect left-right and send left-right buses. All VU meters are also equipped with a peak indicator light, which comes on at roughly 8 dB below clipping.

INTERFACING WITH THE OUTSIDE WORLD

Part of our plan for making the 8428 simple to operate was to place as much signal control as possible right on the board, thereby, eliminating the need for many external control devices. This is accomplished in part by two banks of switches located at the far right and far left of the console.

On the far left, there are six pair of switches which are attached to multipin connectors on the back panel of the console. These provide a series of dry contacts for on/off control of external devices. You can control up to six devices: tape machines, turntables, studio lights, etc.

On the far right, there are another six switches. These are used to address the four group outputs to the multitrack machine through the multipin connector dedicated for that purpose. They essentially eliminate the need for a patch bay to get the signal out to the multitrack. For more sophisticated applications, of course, the customer may still require a patch bay. But for more basic applications, the user is no longer compelled to go out and buy a patch bay just to address the tape machine.

Each of the six switches can route all four output groups to a different set of tracks. Punch the 1-4 switch and groups 1, 2, 3 and 4 are sent to tracks 1, 2, 3 and 4. Punch the 5-8 switch and the 4 groups are sent to tracks 5, 6, 7 and 8, and so on, up to 24 tracks. You can select more than one switch at a time. So if, for example, you want to put signals on tracks 1 and 6, it's no problem. Simply punch up both groups of tracks and then only enable the appropriate record inputs on the multitrack machine itself.

LOOKING TOWARD THE FUTURE

By furnishing a sound basis for the WR-8428, the earlier WR-8616 post-production console laid the groundwork for what may well become a full line of Ramsa post-production consoles. With the 8428, we feel we have been able to zero in on the post-production area, but provide flexibility for all the different types of applications within that area-- broadcast, audio-for-video, radio, industrial users, etc. Apart from the Dolby matrix module, other optional modules for the 8428 may follow in the future, providing even more flexibility. We look forward to a healthy and growing relationship with the post-production community. ■

db Buyer's Guide

CONSOLES

ALLEN & HEATH BRENELL

The SR Series is a full line of sound reinforcement consoles available in 2 and 4 bus versions. In a stereo configuration, it is available as an 8 x 2 x 1 (SR-8), 12 x 2 x 1 (SR-12), and 16 x 2 x 1 (SR-16). In a 4 bus version, which is also designed for 4-track recording applications, it is available in 16 x 4 x 2 x 1 (SR-416), 24 x 4 x 2 x 1 (SR-424), and 32 x 4 x 2 x 1 (SR-432). All models feature 4-band eq, 4 aux sends, and 100 mm faders.

The System 8 Series is designed for 8- and 16-track recording, as well as sound reinforcement applications. Primary output levels are switchable between +4 and -10 operating levels. It is available in a 16 input configuration (Model 1616 Mk II), and the 24 input Model 2416.

The CMC Series are multitrack recording consoles featuring 16 mix buses for use with 8, 16, and 24-track recording. Microprocessor control of bus assignment and muting offer 3 levels of control, internal control with an on-board computer allows for 32 presets and control from a front panel keyboard or external footswitch.

The Sigma Series is a 24-bus series of consoles designed for 16 or 24-track recording applications, and sound reinforcement. Standard modules feature 4-band sweepable eq and 6 aux sends. PFL, AFL, and solo in place monitoring systems are incorporated on all modules, as well as local muting and microprocessor controlled remote muting.

AMEK

The BCII is a small frame audio console for either portable broadcast or edit bay type applications. It is available with a large variety of input and output modules. Modules available are mono mic/line, stereo line, mono or stereo groups, and mono or stereo outputs.
Price: $10,000.00-30,000.00 depending on configuration.

The TAC SR9000 is a sound reinforcement console. Standard format is 40 inputs with 16 mono sub-groups, a stereo output bus, 16 x 8 output matrix, 8 independent mute groups, 8 optional VCA groups, and a total of 16 aux sends.
Price: Starting at $40,000.00.

The TAC Matchless is a full 24-bus/24-track console that is in-line monitor type. Features include 4-band eq with selectable high and low frequency points, swept mids with Q divide control, 2 programmable mute groups and 8 aux sends.
Price: $25,000.00.

The TAC Scorpion is available in a wide variety of frame sizes, configurations, and module types. It is suitable for both recording and sound reinforcement. Frame sizes are 8, 16, 24, or 32-tracks, with input configurations of 16, 24, 32 or 40 inputs.
Price: $7,000.00-20,000.00, depending on options and configurations.

The APC1000 is available in standard formats of 32, 48, 64, 80, 96 and 120 inputs. Features include Recall, GLM Moving Fader System, and complete Plasma metering. Console is very narrow due to many switch functions being put into software enabling resetting from SMPTE.

The G2520 is available in standard formats of 56 and 40 inputs. Integral digital grouping cards allow fitting either MasterMix or Arms at low additional cost. GLM Moving Fader Automation is also available. Standard is Plasma metering and in-line monitor.
Price: $100,000.00-200,000.00.

The Angela is available in standard formats including 28, 36 and 51 input chassis. It is in-line monitor type with full 24- or 48-track routing available. Features include 4-band eq with swept mids and Q divide, high and low pass filters, and extensive patchfield facilities.
Price: $40,000.00-90,000.00.

The Classic is a general purpose console primarily for use in broadcast production applications. It features 4-band parametric eq and sweep filters per module, 8 buses, 2 stereo buses, optional VCA faders with DC sub-grouping, and small width.
Price: $40,000.00-90,000.00.

AMR

The AMR 42 multitrack mixing console has 4 inputs with 4-track and stereo outputs, high impedance mic inputs, mic/line/tape switching, mic/line input, peak reading LED, output level indicators, graphic eq, and internal headphone amplifier.
Price: $349.50.

The AMR 64 is a multitrack mixing console with 6 inputs, 4 channel and stereo outputs, 28 dB headroom, separate monitor and mixer sections, sweepable eq on each input, XLR mic inputs, aux sends, and peak reading LED.
Price: $599.50.

The AMR System I is a multitrack recording system utilizing the MCR-4 and a mixing console with 4 inputs feeding 4-track and stereo outputs. Each input channel features high impedance 1/4-in. phone type mic input, mic/line tape switch, straight line fader and 2-band eq.
Price: $1,398.50.

The System II is a multitrack recording system utilizing the MCR-4 and a 6-input mixing console that feeds 4 channel and stereo outputs. It has separate monitor mixer section to use for overdubbing, input channel patching, and XLR mic inputs.
Price: $1,139.50.

AUDIO-TECHNICA

The AT-RMX64 is a 4-track, 3.75 in./sec. cassette recorder/mixer. It has Dolby B and C, 6 mic/line input channels with 2-band parametric eq, 4 output buses, 2 send/return loops, +/-15% speed control and transformerless inputs and outputs with +4 dBm output.
Price: $1,695.00.

CARVIN

The MX-2488 is a full function 24-channel recording console. It features 24 input channels, and 8 main output channels, and is designed to serve as the control center for an 8-track studio. It includes independent control room monitoring, bus/tape source selection, and 3-band parametric eq.
Price: $3,995.00. (Also available in 16-channel version: $2,995.00.)

The MX-1644 is a full function recording console with 16 input channels, and 4 main outputs. It is a production quality console designed to serve as the control center of a 4-track studio. It features mic/line switching, 4-band fixed eq, and built-in talkback.
Price: $1,695.00.

The MX-22 is available in 6, 8, 12, 16, and 24-channel formats. It is designed for live sound reinforcement or 2-track recording. It features ultra-low noise pre-amps, 48 volt phantom powering, 3-band eq with sweepable mid-range, and 2 monitor sends per channel.
Price: From $849.00-$1,899.00.

The CP-600 and CP-630 feature 6 channels and monaural powered outputs. Both models are identical in features and range from 150 watts to 300 watts. They offer volume, bass, treble, and reverb/effects bus per channel. The outputs are capable of driving a tape deck, or high powered speaker system.
Price: $399.00-$469.00.

DOD

The R-855 is a rack mounted, 4-input, stereo output mixer, that is one rack space high, with pan pots, headphone output, master level control, effects send/receive loop, and a clipping indicator. The mixer can be used with mic, line or instrument levels.
Price: $269.95, (XLR version: $299.95.)

ELECTRO-VOICE

The 8400 Series is available with 8 (8408), 16 (8416), 24 (8424), or 32 (8432) input channels, and stereo tape input. They feature 4 sub-groups, balanced inputs and outputs, stereo main out with independent mono sum, and 3-band eq.
Price: 8408, $3,210.00; 8416, $4,185.00; 8424, $6,160.00; 8432, $7,700.00.

The 8200 Series is available with 8, 12, or 16 input channels, balanced inputs and outputs, stereo main out with independent mono sum, monitor send, 2 aux sends, 3-band eq, and input and output solo.
Price: 8208, $2,140.00; 8212, $2,580.00; 8216, $3,165.00.

The EVT 5200 Series II is available with 8, 12, or 16 input channel. It features balanced inputs and unbalanced outputs, 3-band eq, monitor send and effect send, stereo output with independent mono sum, and 48 volt phantom powering.
Price: 8208II, $990.00; 5212II, $1,195.00, 5216II, $1,530.00.

FENDER

The 3216 powered mixer has 16 inputs with left/right and monitor 1/monitor 2 outputs. Each output has a 9-band graphic eq. Features include 3-band eq, spring reverb, phantom power, VU meters and power amp limiter.

The Dynamix Series II is a sound reinforcement/recording mixer available with 6, 12, or 16 inputs, with stereo outputs. It has 3-band eq, 2 aux buses with returns, and external power supply.
Price: $549.00 for 6 x 2; $849.00 for 12 x 2; and $1,149.00 for 16 x 2.

FOSTEX

The 450 8-track recording mixer has a parametric eq, 8 inputs, 4-channel bus, stereo bus, mono bus, switchable phantom power on each channel, and in-line monitoring.
Price: $1,095.00.

The 450-16 is a 16-track recording mixer with parametric eq, 4-channel bus, stereo, mono, and solo buses, in-line monitoring, and switchable phantom powering on each channel.
Price: $1,895.00.

The 2050 is a 10 x 2 line level mixer with gain and pan for each channel, headphone jack with volume control, front panel priority jacks, and rack mountability.
Price: $260.00.

The 260 Multitracker is a 4-track cassette mixer/recorder with 6 inputs. It has independent stereo buses, 2 mono buses, 3.75 in./sec. tape speed, Dolby C, parametric eq, and true rolling punch ins.
Price: $995.00.

The 160 Multitracker is a 4-track cassette mixer/recorder with 4 channel simultaneous recording and accessory patch points.
Price: $695.00.

The X-15 Multitracker is a 4-track cassette mixer/recorder that runs on AC or battery pack. It has 4 independent tape outputs, independent headphone level control, independent bus and treble controls for each main channel, and Monomix section with gain and pan controls for each track.
Price: $350.00.

The 250 AV Multitracker is a 4-track cassette mixer/recorder specifically designed for slide show synchronization. It allows for 4-track simultaneous recording.
Price: $995.00.

GALAXY AUDIO

The Model M802RM is a rack mountable sound reinforcement console with 8 inputs, 2 outputs, and 2 sub-groups. Each channel has 3-band eq, effect send, monitor send, peak indicator, input attenuator, and sub-group panning. Dimensions are 19 x 15 x 4; weight is 25 lbs.

The Model M802TT is a sound reinforcement console with 8 inputs, 2 outputs, and 2 sub-groups. Each channel has 3-band eq, effect send, monitor send, peak indicator, input attenuator, and sub-group panning. Dimensions are 18.5 x 16 x 9.5; weight is 25 lbs.

The Model M1202 is a sound reinforcement console with 12 inputs, 2 outputs, and 2 sub-groups. Each channel has 3-band eq, effect send, monitor send, peak indicator, input attenuator, and sub-group panning. Dimensions are 25.5 x 19 x 5.25; weight is 35 lbs.

The Model M1602 is a sound reinforcement console with 16 inputs, 2 outputs, and 2 sub-groups. Each channel has 3-band eq, effect send, monitor send, peak indicator, input attenuator, and sub-group panning. Dimensions are 31.5 x 19 x 5.25; weight is 37 lbs.

HILL AUDIO

The Promix is a sound reinforcement console available in 24 x 8 x 2 x 1, and 32 x 8 x 2 x 1 configurations. It has 4-band eq with sweepable notch filter on each input, 6 aux sends, 8 aux returns, 8 groups, mono out, 8 x 4 matrix outputs, and VU metering.
Price: $5,999.00 for 24 x 8 x 2 x 1 with 8 x 4 matrix.

The Remix is a recording console available in 16 x 8 x 16 x 2 x 1, and 24 x 8 x 16 x 2 x 1 formats. It has 4-band eq with sweepable notch filter on each input, 6 aux sends, 8 subgroups, and 16-track monitoring with aux sends.
Price: $6,999.00 for 24 x 8 x 16 x 2 x 1 format.

The Multimix is a recording, sound reinforcement and broadcast console available in 16 x 4 x 2 x 1 configuration. It is a rack mounted unit that occupies 8 rack spaces. It has 3-band eq, 2 aux sends, 16 input channels with mic, line and RIAA inputs, and mono out.
Price: $2,099.00.

The Stagemix is a sound reinforcement/monitor console that occupies 8 rack spaces. It has 12 input channels featuring parallel input on XLR transformers, 6 outputs with transformer isolation, and send return patch points.
Price: $2,299.00.

The Rakmix is a sound reinforcement/broadcast console that occupies 14 rack spaces. It has 4-band eq, 4 aux sends, 4 aux returns, send and return patch points throughout, direct outputs, and 12-way LED displays.
Price: $2,559.00.

The Soundmix is a sound reinforcement console with 4-band eq, 4 aux sends, 4 aux returns, send and return patch points throughout, balanced and unbalanced outputs, direct outputs, and 100 mm smooth action faders.
Price: $4,150.00.

MITSUBISHI PRO AUDIO GROUP

The Superstar Recording Console features dual in-line I/O modules, 44 to 84 inputs, 64 centrally assigned mixing buses, 8 to 16 aux send buses, 2 stereo outputs, modular bolt together aircraft frame, optional meter overbridge for peripheral and in-line devices, and Compumix automation.
Price: From $126,000.00-318,000.00.

The Westar 8000 recording Console features dual in-line I/O modules, and 20 to 52 inputs. It has 24 mixing buses, 8 aux send buses, 2 stereo outputs, VU, peak, or 60-segment LED bar graph meters, selectable top panel plug-in eq, and Compumix automation.
Price: From $47,000.00-198,000.00.

The Westar 8100 Broadcast Production Console features dual in-line mono mic/line I/O modules. It has stereo line input modules, stereo sub master modules, stereo output modules, 24 to 60 inputs, 24 multitrack buses, 8 stereo sub master buses, and 2 stereo output buses.
Price: From $57,000.00-168,000.00.

The Westar 8200 Broadcast Post-Production Console features dual in-line mono mic/line I/O modules and from 20 to 52 inputs. It has stereo line input modules, 16 multitrack mixing buses, 4 stereo sub master buses, 2 stereo output buses, 8 aux send buses, and Compumix automation.
Price: From $54,000.00-215,000.00.

The Superstar Broadcast Post-Production Console features dual in-line mono mic/line I/O modules and from 44 to 84 inputs. It has 64 centrally assigned multitrack mixing buses, 8 stereo sub master buses, 2 stereo output buses, and Compumix automation.
Price: From $140,000.00-340,000.00.

The Westar 8300 Film Re-Recording Console has from 16 to 72 inputs, and 8, 16, or 24 mixing buses. It has 10 aux send buses, 3-channel pan bus, film monitor system with 8 x 4 to 24 x 8 matrix selection, and Compumix automation.
Price: From $70,000.00-600,000.00.

NEVE

The 51 Series Stereo Broadcast Consoles are available in 4 standard configurations with Neve Formant Spectrum EQ, channel dynamics, and custom configurations.

The 542 Range are portable and table top consoles for broadcast, remote recording, or one-inch video production. It is available with VCA automation.

The V Series Master Recording Console feature a 48 bus, up to 72-input mainframe. It has in-line limiter/compressor plus separate gate with external key, Neve Formant Spectrum EQ, 8 aux feeds, mono or stereo operation with 56 aux sends on mixdown, and equalized cue mix.

The 8232 is a 32-channel, 24-bus multitrack recording console for music recording, audio-for-video post-production, or film re-recording. It has central output assignment with Instant Reset plus stereo effects returns.

PEAVEY

The Mark IV is a sound reinforcement console available in 16 x 4 x 1 and 24 x 4 x 1 configurations. It features flight-case construction, transformer balanced inputs, pre and post eq channel in/out patch, 4-band eq, PFL, 2 monitor sends, effect send, and on-board reverb.
Price: $2,499.00 for 16-channel; $2,999.00 for 24-channel.

The MD Monitor is a sound reinforcement monitor console with a 16 x 6 configuration. It has special bridging in/out XLR arrangement to eliminate splitter box, balanced inputs, in/out patch points, 3-band eq, and 6 control matrix.
Price: $999.50.

The Mark IV Monitor is a sound reinforcement monitor console with a 24 x 8 configuration. It has special bridging, in/out XLR arrangement to eliminate splitter box, transformer balanced inputs, LED status display, 4-band eq, mute, phase reverse, PFL, and 8 control matrix.
Price: $3,299.00.

The 701R is a sound reinforcement/keyboard mixer with a 7 x 2 x 1 configuration. It has XLR balanced inputs, 4-band eq, monitor send, effect send, pan and level controls, and reverb.
Price: $649.50.

The MD II is a sound reinforcement mixer available in 8 x 2 x 1, 16 x 2 x 1, and 12 x 2 x 1 configurations. Features include balanced XLR inputs, pre eq in/out patch points, 3-band eq with sweepable mid, limit LED, monitor send, and effect send.
Price: $749.50 for 8 channels; $949.50 for 12 channels; $1,149.50 for 16 channels.

The MD IIB is the same as the MD II, but transformer balanced outputs are included for left, right, master and monitor outputs.
Price: From $999.50-1,199.50.

The MS Mixers are available 12 x 2 x 1 and 16 x 2 x 1 configurations. Features include balanced XLR inputs, 3-band eq with sweepable mids, 100 mm faders, channel status LEDs, PFL, 2 monitor sends, 2 effect sends, and patch bay.
Price: MS1221, $1,799.50; MS 1621, $2,099.50.

The Mark III is a sound reinforcement console available in 16 x 2 x 1 and 24 x 2 x 1 configurations. Both models are constructed in road-ready flight case. Channels feature transformer balanced XLR inputs, pre and post eq in/out patch points, and 4-band eq.
Price: Mark III 16, $1,799.50; Mark III 24, $2,399.50.

RAMSA

The WR-S Series stereo mixing consoles offer 6, 10, or 14 mono inputs, 2 channels with stereo line and phone inputs, 48-volt phantom powering, 3-band sweepable eq, and 3 send circuits for effects and monitoring.
Price: WR-S208, 8 x 2 x 1, $1,295.00; WR-S212, 12 x 2 x 1, $1,795.00; WR-S216, 16 x 2 x 1, $2,095.00.

The WR-8210A recording console has 10 inputs, 4 group outputs, 48-volt phantom power on all input channels, left/right outputs, tape recorder sub mix inputs to monitor tape or echo returns, and LED metering.
Price: $2,160.00.

The WR-8112/WR-8118 are sound reinforcement/recording consoles with 18/12 channel inputs, 4 group outputs, 2 stereo outputs and 1 mono master output. It has 48-volt phantom power, 3-band sweepable midrange eq, and selectable tape monitor.
Price: WR-8112, $2,500.00; WR-8118, $3,400.00.

The WR-8616 is a recording/post-production modular mixing console featuring 16 input channel positions, 4 group output positions, left, right, and mono master outputs, and 4 aux outputs. Metering is 8 LED bargraphs and 6 VU meters.
Price: $4,400.00.

The WR-8428 is a recording/post-production modular mixing console has similar features to the WR-8616. It has 28 input channel positions, 4 matrix output positions, and modular construction.

The WR-T812/820 recording consoles feature 12 and 20 input channels, respectively, with 8 group outputs, left and right master outputs, 4 aux sends, 3-band sweepable eq, and tape and signal monitoring. Optional meter bridges available.
Price: WR-T812, $4,040.00; WR-T820, $5,500.00.

ROLAND

The CPM-120 is a sound reinforcement 8 x 2 stereo powered mixer (60 watts x 2 at 8 ohms). Each channel features input attenuation, high and low eq, effect level, pan pot and level controls. The output section features a 7-segment LED array, left and right output controls, and a headphone jack.
Price: $775.00.

The Boss BX-800 is a 8-channel stereo mixer that features 8 high impedance inputs, 2 high impedance outputs and effect send/return jacks. Each channel has input gain, high and low eq, effect level, pan pot and slider level controls. It weights under 5 lbs.
Price: $395.00.

The Boss BX-600 is a compact 6-channel stereo instrument/line mixer that features 6 high impedance inputs, 2 high impedance outputs and peak LED indicator. Each channel has input gain, effect level, pan and level controls. It also has stereo effect returns.
Price: $240.00.

The Boss BX-400 is a keyboard mixer with 4 mono channels featuring a 3-position input switch and a level control. It weighs less than 3 pounds.
Price: $175.00.

The Boss KM-04 is a 4 x 1 instrument mixer for use in multiple instrument set ups. The battery powered unit is also useful as a pre-amp for amplifying acoustic guitar or bass pickups.
Price: $75.00.

ROSS

The R6M sound reinforcement mixer fits in a single rack space and has 6 input channels, 4 transformer balanced, and 2-band master eq.
Price: $149.95.

The R6L sound reinforcement mixer fits in a single rack space and has 6 input channels (high impedance mic), and 2-band master eq.
Price: $99.95.

The PC8 is a powered (190 watts at 4 ohms) mixing console with 8 input channels, 3-band input eq, built in reverb (pan), and 2-band master eq.
Price: $899.95.

The PC12 is a 12-input version of the PC8, with a 3-band master eq.
Price: $1,199.95.

The MPX820 is a MIDI controllable, fully programmable 8-channel mixer with 99 internal memories, fade time from 40 milliseconds to 40 seconds, 3-band eq, 1 monitor send and 1 effect send.
Price: $2,495.00.

The PC4100CP is a powered mixing console with 4 input channels, 2-band eq, XLR and 1/4-inch inputs, built-in reverb, and built-in cassette player.
Price: $449.95.

The RX Series sound reinforcement consoles have 8, 12, or 16 inputs, stereo program out, sum output, 2 effects sends, and 4 channel input expanders available.
Price: From $599.95 to $999.95.

The Series 2000 8 x 4 recording console has 8 input channels, 4 sub-groups, 4-band eq, and 8-track tape monitoring.
Price: $1,299.95.

SHURE

The FP31 audio mixer has 3 inputs, 2 outputs, switchable low-cut filters, and built-in slate microphone. Dimensions are 2 x 6.3 x 5.3; weight is 21.2 lbs.
Price: $900.00.

The FP32 audio mixer has 3 inputs, 2 transformer-coupled outputs, built-in slate microphone, and built-in tone oscillator. Dimensions are 2.3 x 7.25 x 6; weight is 2.5 lbs.
Price: $1,300.00.

The FP42 is a stereo audio mixer with 4 balanced inputs, 2 outputs, low-frequency roll-off switch, center-detent pan pot, battery or AC operation, built-in stereo peak limiters, phantom power, and tone oscillator. Weight is 6.5 lbs.
Price: $750.00.

The M267 microphone mixer has peak program limiter, simplex power, built-in battery pack, LED peak indicator, headphone level control, automatic muting circuit, active gain controls, and transformer balanced inputs and outputs. Dimensions are 3 x 12.2 x 9; weight is 5.2 lbs.
Price: $475.00.

The M268 microphone mixer has mix bus, simplex power, automatic muting circuit, active gain controls, 4 transformer-coupled low-impedance balanced line mic inputs, and 4 high impedance phone jack inputs. Dimensions are 3 x 12.2 x 9; weight is 4.1 lbs.
Price: $257.00.

SECK

The SECK 62 is a portable stereo console for sound reinforcement and recording. It has balanced mic/line inputs with insert point, 3-band eq with sweepable midrange, 4 aux sends, and full-throw 100 mm faders. Dimensions are 14.5 x 18.1 x 2.1.
Price: $1,345.00.

The SECK 122 is the same as the 62, above, but it has 12 input channels. Dimensions are 23 x 18.1 x 2.1.
Price: $1,995.00.

The SECK 242 is the same as the 62, but it has 24 input channels. Dimensions are 40 x 18.1 x 2.1.
Price: $3,450.00.

The SECK 1882 is a portable recording or sound reinforcement console with 18 balanced inputs has mic in line and tape inputs with 3-band eq with sweepable midrange, 6 aux sends, 4 aux returns. Dimensions are 29.5 x 18.1 x 2.1.
Price: $3,995.00.

The SECK 1282 is a 12 input version of the 1882. Dimensions are 39 x 18.2 x 2.2.
Price: $3,450.00.

SOLID STATE LOGIC

The SL 4000E Series consoles are for multitrack recording and mixing and are available with 24 to 64 input/output channels, each with built-in compressor/limiter, expander/gate, high and low pass filters, 4-band parametric eq, 6 cue/aux sends, pushbutton signal processor routing and 32 multitrack group outputs.

The SL 6000E Series consoles are for audio for video post-production and is available in versions accepting up to 64 input/output modules. All versions feature 3 stereo mix buses, patch-free audio sub grouping, and extensive signal processing routing features.

The SL 5000M Series audio production system is a new generation of audio console architecture from which an almost infinite variety of specialized and general broadcast audio consoles may be constructed. Options include Total Recall, Events Controller, and Real Time System.

SONY

The MXP-3000 Series audio recording and remix consoles have modular I/O channel strips, choice of 5 different plug-in equalizers, 6 sends, 12 dB/octave high pass filter, stereo solo monitor, and 2-stage peak indicators.

The MXP-2000 Series broadcast and post-production consoles have 2 separate stereo program feeds, programmable mutes, complete rehearse/on air switching functions, built-in PFL with speaker, up to 32 inputs, and 3-band eq on each input.

The MXP-29 is an 8-channel mixer featuring 4-position input selector on each channel, 48 volt phantom powering, trim control, low cut filter, 3-band eq, 2-channel sub inputs, PFL, and built-in 1 kHz oscillator.

SOUNDCRAFT

The 200SR series sound reinforcement mixing consoles have 4-band fixed eq, balanced mic/line inputs, 4 aux sends, 4 group assignments, 4 effects returns with 2-band eq, 48 volt phantom power, individual channel insert points, balanced mix outputs and VU meters.
Price: From $2,295.00 to $5,175.00.

The 600 Series sound reinforcement mixing consoles have 4-band high and low mid sweepable eq, high pass filter, balanced mic/line inputs, 6 aux sends (pre/post eq fader internal selection), 16 monitor/effects returns with 2-band eq, 48 volt phantom powering and 2 VU meters.
Price: From $9,150.00 to $14,500.00.

The Series Four sound reinforcement consoles have balanced mic/line inputs, 48 volt phantom power, phase switch, variable high and low pass filter, 4-band fully parametric eq, 8 aux sends pre/post fader, and 8 effects returns with input level. The monitor section features 16 post fader sends to output modules, 3-band fully parametric eq, and balanced outputs.
Price: From $50,750.00 to $59,750.00.

The 200 Series recording console features 4-band fixed eq, balanced mic/line input, 4 aux sends, 4 sub group and direct and mix assignment, 8 monitor/effects returns, -10/+4 internal switching, balanced outputs, 48 volt phantom power, and individual channel insert points.
Price: From $2,750.00 to $7,550.00.

The 1600 Series recording console features separate mic and line gain control, 4-band high and low mid sweepable eq, high pass filter, balanced mic inputs, 8 aux sends, 8 sub groups and direct mix assignments, 16-track monitoring, balanced mic and aux outputs, and 48 volt phantom power.
Price: From $14,250.00 to $28,950.00

The 2400 Series recording consoles feature separate mic and line gain control, phase switch, variable high pass filter, 4-band high and low mid sweepable eq, selectable shelf frequency on high and low frequency eq, 6 aux sends pre/post fader, 24-track assignment, solo in place, and A & B mute buses.
Price: From $27,450.00 to $40,950.00.

The TS12 Series recording consoles feature high impedance line input, balanced mic and line inputs, 48 volt phantom power, phase switch, variable high pass filter, 6 aux sends, 2 programmable mute groups, 12 group output section, and 4-band parametric eq.

The TS24 Series recording consoles feature high impedance line input, balanced mic and line inputs, 48 volt phantom power, phase switch, variable high pass filter, 6 aux sends, 24-track assignment, 24-track monitor, parametric 4-band eq, LED meter bridge, and Penny & Giles faders.
Price: From $48,950.00 to $84,250.00.

SOUNDTECH

The SL-16 is a 16 x 2, 4 aux sound reinforcement console with isolated power supply. It features 2 balanced inputs, 1 balanced output per channel, and 2 effect sends. Dimensions are 30 x 24 x 4; weight is 40 lbs.
Price: $1,999.00.

The SL-24 is a 24-channel version of the SL-16, above. Dimensions are 40 x 24 x 4; weight is 49 lbs.
Price: $2,599.00.

The Studio 168 is a 16 x 8 x 2 recording console that has 4 effect buses, balanced inputs and outputs, and 4-band eq. It also has phantom powering, phase reverse, and 16 tape returns. Dimensions are 32 x 24 x 10; weight is 70 lbs.
Price: $5,299.00.

The Studio 248 is a 24-channel version of the 168, above. Dimensions are 44 x 26 x 10; weight is 92 lbs.
Price: $6,999.00.

STUDER REVOX

The Studer 900 Series is a full size production console with frame sizes for up to 56 inputs and 24 mixing buses. Features include 5 pre or post send/returns, 4-band parametric eq, VCA fader option, and transformerless input option. Dimensions for 16-module frame are 29.5 x 40 x 37.
Price: From $36,650.00.

The Studer 961/962 are compact consoles for broadcast, remote recording, and video editing. Frame sizes are for 14 to 20 modules. Features include stereo line inputs with or without 3-band eq, compressor/limiter on master modules, electronic muting, FET switching, and balanced insert points. Dimensions for 961 portable in case are 20 x 12 x 21.3; weight is 55 lbs.
Price: From $11,500.00.

The Studer 963 is a compact production console based on a 30 mm module width. It is available with up to 40 inputs and 8 mixing buses. Features include 3-band eq, 4 switchable pre or post send/returns, external mute interface, direct outputs for each channel, patch bay, and electronic switching. Dimensions are 63.6 x 40.7 x 33 for the 28 x 8 x 4 configuration.
Price: $52,000.00.

SUNN

The SPL 4424 is a sound reinforcement console with 24 inputs with 4 sub-groups, left/right and mono outputs, high pass filter, 4-band eq with selectable mids, 4 aux buses, phantom power and VU meters.
Price: $6,895.00.

The SPL 2216 is a sound reinforcement console with 12 or 16 inputs, sub 1, sub 2, monitor, and main outputs, 4-band eq, 1 effect/reverb bus.
Price: $1,829.00, for 16-channel; $1,479.00 for 12-channel.

TASCAM

The M-600 series mixers are 24 to 32 channel mixers with 16 or 32 channel monitor. It has 8 aux mixes to enable elaborate effects handling. Mono or stereo strips are available.

The Porta-Two is a 6-channel mixer with 4-track cassette recorder. It is battery or AC operated, with tape sync functions for use with MIDI instruments, and effects submixer with stereo returns.

TEK TRAK

The G6.2 stereo mixer features electronically balanced inputs, peak indicators on each channel, 2 large VU meters, Alps faders, 3-band eq, and rack mountability. Dimensions are 17 x 15 x 3; weight is 15 lbs.
Price: $575.00.

The G6.2S stereo mixer is the same as the G6.2, above, but it also has a built-in MOS-FET power amplifier. Dimensions are 17 x 15 x 8; weight is 30 lbs.
Price: $1,050.00.

The GMA200 is a mixer with a 200 watt MOS-FET amplifier that features 8 input channels (mic/line), 3-band eq, 2 aux sends per channel, 10-band graphic eq, analog echo, and tape returns. Dimensions are 20.5 x 17.5 x 6; weight is 35 lbs.
Price: $1,440.00.

The GB10.2 stereo mixer features electronically balanced inputs, (mic/line), peak indicators on each channel, 2 large VU meters, Alps faders, 3-band eq, solo switches, and headphone jack. Dimensions are 22 x 21 x 5; weight is 29 lbs.
Price: $959.00.

The GB16.2 stereo mixer has the same features as the GB10.2, and it is also available with built-in flight case. Dimensions are 30 x 21 x 5; weight is 38 lbs.
Price: $1,490.00, with flight case.

The GA16.4.2 is a stereo mixing console that features electronically balanced inputs (mic/line), 3-band eq with parametric mid sweep, 3 aux sends per channel, switchable meters, external power supply, 2 separate stereo effects return, Alps faders, talkback, and 4-band eq master.
Price: $2,675.00.

The GA24.4.2 is a 24-channel version of the GA16.4.2.
Price: $3,465.00.

The GA16.8.2 stereo mixing console features 16 balanced inputs, 4-band eq, 3 aux sends, mute and solo switches, mic pad and mic line switches, Alps faders, 8 in/out groups each with send/return, 2-band eq, 2 aux sends, and pan control. Dimensions are 36 x 33 x 8; weight is 75 lbs.
Price: $3,815.00.

TRIDENT

The Series 80B console has 4-band high and low mid sweepable eq with high and low frequency 2-position shelving and 50 Hz, 12 dB/octave high pass filter. It has 24 separate bus outputs and direct mix assignments, and channel mute with channel AFL and PFL. It is available in frame sizes of 30, 40, and 50 inputs.
Price: 30 x 24 x 24, $49,950.00; 40 x 24 x 24, $69,950.00; and 50 x 24 x 24, $89,950.00

The Series 65-4 consoles have 4-band high and low mid sweepable eq with variable high pass filter, balanced mic and line inputs with separate gain controls and phase reverse, 8 aux sends with pre/post switching, 8 monitor/effects returns, and stereo in place solo.

The Series 75 consoles have 24 bus outputs and 24 monitor/effects returns, 4-band high and low mid sweepable eq with variable high pass filter, balanced mic and line inputs with separate gain controls and phase reverse, 8 aux sends, and 24 sub-groups with direct mix assignments. It is available in 28, 36, and 40 input frames.

BUYERS GUIDE ADDRESS LISTS

CONSOLES & MIXERS

Allen & Heath Brennell USA Ltd.
5 Connair Drive
Orange,CT 06477

Amek Consoles, Inc.
10815 Burbank Blvd.
North Hollywood,CA 91601

AMR
Route 2, Highway 503
Decatur,MS 39327

Audio Technica
1221 Commerce Dr.
Stow,OH 44224

Carvin Mfg. Corp.
1155 Industrial Ave
Escondido,CA 92025

Connectronics
652 Glenbrook Rd.
Stamford,CT 06906

Digital Entertainment Corp.
225 Parkside Dr.
San Fernando,CA 91340

DOD Electronics
5639 So. Riley Lane
Salt Lake City,UT 84107

Electro-Voice
600 Cecil St.
Buchanan,MI 49107

Fender
1130 Columbia St.
Brea,CA 92621

Fostex
15431 Blackburn Ave.
Norwalk,CA 9065f0

Galaxy Audio
625 East Pawnee
Wichita,KS 67211

Hill Audio,Inc.
5002 N. Royal Atlanta Dr. #B
Tucker,GA 30084

Mitsubishi
(see Digital Entertainment Corp.)

Rupert Neve,Inc.
Berkshire Industrial Park
Bethel,CT 06801

Peavey
711 A Street
Meridian,MS 39301

Ramsa
Panasonic Industrial Co.
6550 Katella Ave.
Cypress,CA 90630

RolandCorp US
7200 Dominion Circle
Los Angeles,CA 90040

Ross
1316 E. Lancaster
Ft. Worth,TX 76116

Shure Bros.
222 Hartrey Ave.
Evanston,IL 60204

Seck (see Connectronics)

Solid State Logic Ltd.
200 West 57th St,Suite 1210
New York,NY 10019

Sony
Sony Drive
Park Ridge,NJ 07656

Soundcraft USA
8500 Balboa Blvd.
Northridge,CA 91329

SoundTech (see Washburn Int.)

Studer Revox
1425 Elm Hill Pike
Nashville,TN 37210

Sunn (see Fender)

Tascam
Teac Corp. of America
7733 Telegraph Rd.
Montebello,CA 90640

Tek Trak
Rt. 165,Box 431
Voluntown,CT 06384

Trident (USA) Inc.
308 N. Stanley Ave.
Los Angeles,CA 90036

Washburn International
230 Lexington Dr.
Buffalo Grove,IL 60090

db Buyer's Guide

TAPE AND TAPE RECORDERS

AMR

The MCR-4 multitrack cassette recorder has zero stop and play function, Dolby B and C, peak hold level indicators for each channel, solenoid operated controls, pitch control, 4-digit timer, front panel headphone jack with level control, and mute switch.
Price: $699.50.

FOSTEX

The E-16 is a synchronizer ready, 16-track tape machine with built-in 2 position autolocator, servo control of reels, spot erase, real-time counter, 15 in./sec. tape speed, 10.5-in. reels, and Dolby C noise reduction. Weight is 78 lbs.
Price: $6,995.00.

The E-8 is an 8-track version of the E-16.
Price: $4,295.00.

The E-2 is a 2-track recorder with center track for stripping SMPTE time code. It has built-in 2 position autolocator, servo controls of reels in edit mode, spot erase, and balanced inputs and outputs.
Price: $3,600.00.

The E-22 is a 2-track recorder with center track for time code. It has built-in 2-position autolocator, servo controls of reels in the edit mode, spot erase, automatic programmable punch in/out, and balanced inputs and outputs.
Price: $3,900.00.

The M-80 is a synchronizer ready 8-track recorder. It utilizes 1/4-inch tape on 7-inch reels, and has a frequency response of 40 Hz-18 kHz at 15 in./sec. Dimensions are 14 x 13.5 x 6.75; weight is 29 lbs.
Price: $1,995.00.

The M-20 is a 2-track recorder with center track for time code. It can be used with all synchronizers and most video editors. Tape speed is 15 and 7.5 in./sec. Dimensions are 14 x 13.5 x 8.5; weight is 29 lbs.
Price: $1,200.00.

NAKAMICHI

The MR-1 professional cassette deck has 3 discrete heads, dual capstan, balanced (+4 dB) operating levels, rack mountability and a frequency response of 20 Hz-20 kHz, +/- 3 dB, S/N ratio of >70 dB, and less than 0.048% wow and flutter.

The MR-2 professional cassette deck is rack mountable and has variable output levels, Dolby B and C, a frequency response of 20 Hz-20 kHz, +/-3 dB, a S/N ratio of >68 dB, and less than 0.11% wow and flutter.

OTARI

The MX-5050 Mark III/8 is a compact, table top console recorder with a 1/2-inch, 8-track format. Proprietary microprocessors govern dynamic braking, motion sensing, and transport logic. Reel size is 10.5 x 1/2-inches.
Price: $5,835.00.

The MTR-10 and MTR-12 Series II are microprocessor controlled recorders designed for recording studio, and audio post production, and broadcast. It is available in 1/4-inch, 2-track; 1/2-inch, 2-track; 1/2-inch, 4-track, and 1/4-inch, 2-track with time code center track.

The MTR-20 is a microprocessor controlled analog mastering machine designed for broadcast, recording, and post-production. It is available in 6 formats, 1/4-in. mono, 1/4-in. 2-track, 1/4-in. 2-track with center channel time code, 1/4-in stereo, 1/2-in. 2-track, and 1/2-inch 4-track.
Price: For 1/4-inch 2-track, $12,650.00.

The MX-70 is a multitrack mastering recorder for audio post-production and recording. It features microprocessor-controlled constant-tension transport, and noiseless and gapless insert recording capability. Reel size is 10.5 x 1-in.
Price: $16,750.00.

The MX-5050 BQ-II is a compact, table-top console recorder with a 1/4-inch, 2-track format. It has optimized 3-head design, and transformerless balanced inputs. Reel size is 10.5 x 1/4-inch.
Price: $3,595.00.

The MX-80 is a 2-inch, multitrack mastering tape machine for audio post-production and recording. It features microprocessor controlled constant tension transport, and noiseless, gapless punch in and punch out. The record circuitry incorporates Dolby HX-Pro.
Price: $34,950.00.

The MTR-90-II is a microprocessor controlled, pinchroller-less master multitrack tape machine available in 1-inch and 2-inch transport configurations. It is designed to easily interface with any video editing system, tape controller or tape synchronizer.
Price: $44,950.00.

The DTR-900 is a 1-inch, 32-track digital audio tape machine based on the PD recording standard and is available in 1-inch, 32-track, or 1-inch 24-track (expandable to 32-tracks) configurations. Reel size is 14 x 1-in.
Price: $189,00.00.

SOUNDCRAFT

The Series 760 MK-3 is a 2-inch, 16 or 24-track tape machine with 15 and 30 in./sec. tape speeds, interchangeable headblock, custom transport with heavy duty DC servo-controlled motor, front panel alignment capability, optimized VU metering, remote punch in/out jack, and optional remote autolocator.
Price: From $19,750.00 to $27,450.00.

The Saturn Multitrack tape machine is totally remote with custom transport switches, multi-function 10-memory autolocate including return to zero/local zero/tail out start, variable speed, push button alignment with test oscillator, 3 tape speeds, and 3 eq curves.

STUDER REVOX

The Revox B77 MkII is a compact professional recorder with die-cast chassis, 3 heads, servo-controlled capstan motor, vari-speed control, and 2 or 4-track stereo. It is available with any 2 adjacent tape speeds up to 15 in./sec. Dimensions are 18 x 16 x 8; weight is 37.5 lbs.
Price: $1,999.00.

The Revox PR99 MkII is a compact professional recorder with balanced and floating inputs and outputs, microprocessor controlled real time counter, return to zero autolocate, loop, tape dump, vari-speed and self-sync. Dimensions are 19 x 15.7 x 8; weight is 40 lbs.
Price: $2,595.00.

The Studer A810 is a professional broadcast recorder with microprocessor control of all transport and audio electronic functions. Features include 4 tape speeds, selectable softkey functions, zero locate, digital alignment, and optional center track time code. Dimensions are 18.3 x 19.2 x 8.9; weight is 66 lbs.
Price: $5,950.00.

The Studer A812 is a professional recorder with 12.5-inch reel capacity, 4 tape speeds, thumbwheel shuttle/edit control, choice of 40 user programmable functions, transformerless in/out, optional console, serial remote, and center-track time code.
Price: $8,450.00 for 2-channel version.

The Studer A80VU MkIV is a multichannel professional recorder in 4, 8, 16, and 24-channel versions. Features include die-cast chassis, modular electronics, master bias oscillator, servo control of capstan and spooling motors, and optional synchronizer interface.
Price: From $12,650.00 to $29,900.00.

The Studer A820 is a 2-channel mastering recorder with microprocessor control of all transport and audio electronics parameters. Features include 14-inch reel capacity, 4 tape speeds, programmable function library, and digital setting of alignment parameters.
Price: From $9,350.00.

The Studer A800 MkIII is a microprocessor controlled multichannel recorder in 8, 16, and 24-channel versions. Features include 14-inch reel capacity, phase linear amplifiers, and master bias control.
Price: From $30,000.00.

TANDBERG

The TCD 910-911 is a 4 motor cassette recorder with built-in oscillators for azimuth, bias and record current alignment, 8-bit microprocessor with 32k EPROM, XLR input/output connectors, and optional RS 232 interface.

TASCAM

The ATR-80 is a 24-track recorder with 2-inch transport, featuring many editing features and extensive interface capabilities.

The 112 is a rack mounted professional cassette recorder with cue function in both fastwind modes, Dolby HX-Pro, and Dolby B and C noise reduction systems.

The 112R is an auto-reverse, rack mounted professional cassette deck with Super Acculign Rotating Head System, and it is capable of multiple deck operation via a 16-pin connector.

The 112C is an update of the 122 professional cassette recorder.

The ATR-60 Series tape recorders includes the ATR-60-2T with center track time code and coincident head configuration; the ATR-60-4HS and 2HS 30 in./sec., 1/2-inch, 4-track and 2-track recorders; the ATR-60-2N 2-track mastering machine; and the ATR-60-8 1/2-inch, 8-track.

UHER

The 1200 Report Synchro is a portable open reel, full-track mono tape recorder with pilot track, 7.5 in./sec. speed, 3 heads, 5-inch reel, belt drive, VU meter, 2 mixable mic inputs, and switchable ALC selectable record/playback eq. Dimensions are 11 x 3.5 x 9; weight is 8 lbs.
Price: $4,819.00.

The 6000 Report Universal is a portable open reel 2-track mono tape recorder with 4 speeds, 3 heads, 5-inch reel, solenoid control, belt drive, 1 VU meter, built-in voice activation system, memory pulse facility, and full remote control. Dimensions are 11 x 3.5 x 9; weight is 8 lbs.
Price: $1,729.00.

The CR 160AV is a portable cassette, 4-track stereo tape recorder. It has 2 heads, 2 VU meters, Dolby B and C, switchable ALC, sync dubbing outlet, LED function indicator, and solenoid control. Dimensions are 9 x 2 x 7; weight is 7 lbs.
Price: $899.00.

The CR 1601 Monitor is a portable cassette, 4-track mono tape recorder with 3 speeds, 3 heads, VU meter, switchable ALC, solenoid control, full remote control and built-in voice activation system. Dimensions are 9 x 2 x 7; weight is 7 lbs.
Price: $1,729.00.

The 4000 Report Monitor AV is a portable open reel, 2-track mono tape recorder with 4 speeds, 3 heads, 5-inch reel, belt drive, 2 VU meters, and switchable ALC. Dimensions are 11 x 3.5 x 9; weight is 8 lbs.
Price: $1,449.00.

The 4200 Report Monitor is a portable open reel, 4-track stereo tape recorder with 4 speeds, 3 heads, 5-inch reel, belt drive, and 2 VU meters. Dimensions are 11 x 3.5 x 9; weight is 8 lbs.
Price: $1,549.00.

The 4400 Report Monitor is a portable open reel, 4-track stereo tape recorder with 4 speeds, 3 heads, 5-inch reel, belt drive, 2 VU meters, LED function indicators, switchable ALC. Dimensions are 11 x 3.5 x 9; weight is 8 lbs.
Price: $1,549.00.

TAPE

AGFA-GEVAERT

The PEM 369 is 1 mil open reel mastering tape that features high output, low noise, low print through, and long play times.

The PEM 469 is studio mastering tape featuring high output and low noise, wide dynamic range, standard bias, low print through, and good winding characteristics.

The PEM 468 is studio mastering tape with high output, low noise, wide dynamic range, low print through, and batch number and web position printed on the backing for permenant tape identification.

The PEM 297D is digital audio mastering tape combining low drop out characteristics, and mechanical stability.

The PE 627/827 are extremely low noise, pure chromium cassette tapes with IEC bias II, 70 microsecond chrome equalization.

The PE 619/819/1219 are low noise, high output iron oxide cassettes.

The Magentite 62/92 is bulk cassette tape with extremely low noise, high output and extended dynamic range. It has improved Magnetite formulation designed for music tape duplication.

The PEM 526 is bin loop tape with high mechanical stability and consistant high frequency reproduction.

AMPEX

Grand Master 456 Studio Mastering tape is analog mastering tape available in 1/4, 1/2, 1, and 2-inch widths, and 1,200 to 5,000-foot lengths. The base film is nominally 1.5 mil polyester with gamma ferric oxide and high conductivity carbon backcoat.

Ampex 406 analog audio mastering tape is available in 1/4, 1/2, 1, and 2-inch widths, and 600 to 5,000-foot lengths. Base film is 1.5 mil polyester with gamma ferric oxide and high conductivity carbon backcoat. 407 mastering tape is the same as 406 but the base film is 1 mil thick, and it is available in 900 to 3,600-foot lengths.

Ampex 467 Digital Mastering tape is available in 1/4, 1/2, and 1-inch widths, and 4,600, 7,200, and 9,700-foot lengths. Base film is 1 mil polyester with a cobalt modified gamma ferric oxide and high conductivity carbon backcoat.

Ampex 467 digital audio cassettes are U-Matic cassettes specifically designed for digital audio PCM applications. The cassettes are available in 30, 60, and 75-minute play lengths. Base film is polyester, with thicknesses of 0.81, 0.75, and 0.57 mils, respectively.

Ampex 600 Series open reel and duplicator tape has a polyester base film in 0.5, 1, and 1.5 mil thicknesses, and 1/4-inch width. Reel configurations are 600 to 3,600-feet, and 2,500 to 7,200 feet for duplicator tape. It utilizes gamma ferric oxide and no backcoat.

Ampex 615/616 cassette duplicator tape is Type I tape for C-60 (615), and C-90 (616) duplication. Base film is polyester in 0.45 and 0.26 mil thicknesses, respectively, and 0.15-inch width. Both use ferric oxide coating and no backcoat.

Ampex 619/620 cassette duplicator tape is Type II extended range tape for C-60 (619) and C-90 (620) duplication. Base film is polyester in 0.44 and 0.28 mil thicknesses, respectively, and 0.15-inch width. Both utilize a chromium dioxide coating and no backcoat.

Ampex 672 professional audio cassettes are available in 30, 45, 60, and 90 minute play lengths. It is available in packaged or bulk configurations. Ampex 615 and 616 tape is used in conjunction with a precision molded cassette shell.

BASF

The LH Extra I cassette tapes utilize high performance ferric tape and are available in C-60, and C-90.
Price: $1.59, and $1.89, respectively.

The LH Maxima I cassette tapes utilize high performance tape with enhanced low and high frequency MOL values. It is available in C-60, and C-90.
Price: $2.19, and $2.79, respectively.

The Chrome Extra II cassette tapes utilize pure chrome tape with extra low and high frequency sensitivity and MOL, and ultra-low bias and modulation noise. It is available in C-60 and C-90 lengths.
Price: $2.89, and $3.59, respectively.

The Chrome Maxima II cassette tapes are high density formulation with enhanced low and high frequency MOL for extra dynamic range. It is available in C-60 and C-90. Price: $4.59, and $4.79, respectively.

The Loop Master 920 open reel tape is chrome mastering tape with back-coated design for high-speed bin mastering use. Dynamic range at 3.75 in./sec. is equal to a ferric master recorded at 7.5 in./sec. It is available in 1/2-inch and 1-inch configurations on 2,400-foot hubs.
Price: $42.00 and $83.00, respectively.

MAXELL

The Communicator series cassettes are available in lengths from C-30 to C-120. Price: Ranges from $1.55 to $3.71.

The Duplicator series cassettes are available in lengths from C-30 to C-120.
Price: Ranges from $1.44 to $3.45.

The XL series 1/4-inch open reel tape is back coated and is available in lengths of 1,800-ft. (90 minutes), 2,500-ft. (2 hours), and 3,600-ft. (3 hours).
Prices: $10.39, $26.79, and $28.99, respectively.

The XLII series 1/4-inch open reel tape is back coated and is available in 1,800-ft. (7-in. reel) and 3,600-ft. (10.5-in. reel) lengths.
Price: $13.79 and $35.79, respectively.

TDK

The MA-R are type 4 cassettes with metal particle formulation has extremely high coercivity and remanace for superior full frequency MOL and low distortion. It features reference standard mechanism and die cast metal alloy frame.

The MA cassettes are the same as the MA-R, but it is encased in a laboratory standard mechanism.

The HX-S is a pro-series metal particle cassette tape requiring type II bias. It has a lab standard mechanism.

The SA-X is a Super Avilyn cassette tape available in C-60 and C-90. It has a lab standard mechanism.

The SA cassettes are the same as the SA-X but it has extended highs.

The AD-X is normal bias type I cassettes featuring Avilyn particles high MOL, greater high and low frequency sensitivity and greater headroom.

The AD-S is Avilyn particle cassettes with normal bias. It features clear C-thru shell mechanism.

The AD is premium normal bias cassettes with standard cassette mechanisms.

The GX-50 is a back-coated, 1/4-inch, open reel 1.5 mil high output, low noise tape available on both 7-in. and 10.5-in. reels. The magnetic material is gamma ferric oxide.

The GX-35 is same as the GX-50, but is is 1-mil tape.

The SA/EE is Super Avilyn particle 1/4-inch tape engineered for 1/2-speed open reel decks featuring EE (extra efficiency) eq/bias position. Coercivity is almost double that of standard ferric oxide tape.

3M (SCOTCH)

The 250 audio mastering tape incorporates a base thickness of 1.5 mils and a polyester backing. It delivers high output/low noise performance and offers a wide dynamic range and strong durability.

The 226 audio mastering tape has a base thickness of 1.5 mils and a polyester backing. It provides high output without distortion and has a smooth surface.

The 227 audio tape combines a 1-mil base with a polyester backing. It is similar to the 250 and 226 in performance characteristics with a longer playing time.

The 275 digital audio tape has a base thickness of 1 mil and a polyester backing. It offers high-quality performance in digital applications.

The 806 audio mastering tape has a base thickness of 1.5 mils and a polyester backing. It was developed using the same oxides as the high-performance 226 mastering tape. It is particularly suited to on-location recording applications.

The 807 audio tape incorporates a base thickness of 1 mil and a polyester backing. It offers the same performance specifications as the 806 tape.

The 808 audio mastering tape combines a base thickness of 1.5 mils and a polyester backing. It offers an extremely low signal to print ratio to provide strong performance.

The 809 audio mastering tape offers the same benefits as the 808, but it has a 1-mil backing to provide longer playing time.

TAPE RECORDERS

AMR
Route 2,Highway 503
Decatur,MS 39327

Fostex
15431 Blackburn Ave.
Norwalk,CA 90650

Nakamichi,USA Corp.
19701 S. Vermont Ave.
Torrance,CA 90502

Otari Corp.
2 Davis Dr.
Belmont,CA 94002

Soundcraft USA
8500 Balboa Blvd.
Northridge,CA 91329

Studer Revox
1425 Elm Hill Pike
Nashville,TN 37210

Tandberg of America
Labriola Ct.
Armonk,NY 10504

Tascam
Teac Corp. of America
7733 Telegraph Rd.
Montebello,CA 90640

Uher of America
7067 Vineland Ave
N. Hollywood,CA 91605

TAPE

Agfa Gevaert
275 North St.
Teterboro, NJ 07608

Ampex Mtd.
401 Broadway
Redwood City,CA 94063

BASF
19 Crosby Drive
Bedford,MA 01730

Maxell Corp. of America
60 Oxford Drive
Moonachie,NJ 07074

TDK
12 Harbor Park Drive
Port Washington,NY 11050

3M/Magnetic Media Division
3M Center-Bldg. 223-5S-01
St. Paul,MN 55144

New Products

NEW MIXING CONSOLES

3rd Generation Professional Audio of England has entered the US market with a new line of stereo mixing consoles and dual channel MOS-FET amplifiers. Top of the list is the GA1682 recording console. This unit has standard recording features, and it can be rigged to accomodate a 24 into 2 configuration for live performances. It is designed to be user friendly, and with roadworthy durability.
Mfr: TEK TRAK
Price: $3,815.00.

Circle 60 on Reader Service

YAMAHA MIDI STUDIO SYSTEM

The MIDI Studio System combines a multitrack recorder/mixer, a sequencer, rhythm programmer and speakers with the DX100 FM digital synthesizer to produce a high-quality personal studio available from a single source. The heart of the system is the performance-oriented DX100, designed for the keyboardist who is new to the world of digital synthesis. It is a comact, portable 49-note keyboard featuring 192 preset voices and a programmable FM tone generator system for creating new voices. Also in the system is the full-featured and compact MT1X 4-track cassette recorder with built-in 4-channel mixer. The mixer section includes Tape/Mic/Line input selectors on each channel, level adjustment for mic and line input, Auxilliary Send and Master Return inputs for effects processing, and a separate monitoring section with Level and Pan controls for each track. It weighs 5.5 lbs. High performance digital sequencing is provided by the QX21 Digital Sequence Recorder, which is capable of recording anything played on a MIDI keyboard, complete with touch sensitivity and function parameters captured exactly as played. It has a capacity of more than 8,000 notes, and resolution is an extremely fine 1/384th of a measure, permitting creation of pieces that would be impossible to play live. The system's heartbeat is provided by the RX21 Digital Rhythm Programmer, which offers advanced computer control over nine live drum sounds using Pulse Code Modulation. Rhythms can also be entered in real-time by playing on the corresponding drum keys or by entering patterns step by step. It has 40 preset patterns and 56 user-programmable patterns. Also in the system is the YMC10 MIDI Converter for converting MIDI signals for tape recording, and the KS10 powered speakers. The KS10 is portable and compact enough to rest on top of a keyboard, and it contains a 6.5 watt amplifier. The studio system will handle everything from recording live sounds and MIDI signals, to mixdown, overdubbing, adding effects, sequencing, and sending program changes, all accomplished using the same recording studio techniques as professionals.
Mfr: Yamaha International Corp.
Price: Approximately $1,800.00.

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People, Places...

• **Unique Recording Studios** recently upgraded its Manhattan, New York facility by purchasing four new **Studer A800** 24-track recorders to replace existing recorders in Studio A and B. Two of the Studers are locked together for 48-track operation using **Adams-Smith** synchronizers.

• **BASF Corporation Information Systems** has appointed **Jeffrey L. Brown** to the position of product manager for professional audio/video products. In his new position, Mr. Brown will report to Larry Rallo, marketing manager for professional products. His duties will include packaging, pricing, forecasting and promotion of the company's line of duplicator and studio products.

• As of September 2, 1986, the New York City field office of **Studer Revox America** has moved to 161 Avenue of the Americas, Suite 901. The telephone number, (212) 255-4462, remains unchanged.

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